The Pan-European Mobile Radio System Part I

L. Hanzo, R. Steele

Department of Electronics and Computer Science, University of Southampton

Highfield S09 5NH Southampton - UK

Abstract. Following the launch of the Pan-European digital mobile radio (GSM) system its salient features are summarised in this tutorial review [1, 8]. Time Division Multiple Access (TDMA) with eight users per carrier is used at a multi-user rate of 271 kbit/s, demanding a channel equaliser to combat dispersion. The error protected chip-rate of the full-rate traffic channels is 22.8 kbit/s, while in half-rate channels is 11.4 kbit/s. There are two speech traffic channels, five different-rate data traffic channels and 14 various control and signalling channels to support the system's operation. A moderately complex, 13 kbit/s Regular Pulse Excited speech codec with long term predictor (LTP) is used, combined with an embedded three-class error correction codec and multi-layer interleaving to provide sensitivity-matched unequal error protection for the speech bits.An overall speech delay of 57.5 ms is maintained. Slow frequency hopping at 217 hops/s yields substantial performance gains for slowly moving pedestrians.

1. INTRODUCTION

Following the standardisation and launch of the Pan-European digital mobile radio system known as GSM it is of practical merit to provide a rudimentary introduction to the system's main features. The GSM specifications were released as 13 sets of Recommendations [10], which are summarised in Table 1, covering various aspects of the system [R.01.01].

Table 1 - *GSM Recommendations [R.01.01]*

R.00 - Preamble to the GSM Recommendations.

R.01 - General structure of the Recommendations, description of a GSM network, associated recommendations, vocabulary, etc.

R.02 - Service Aspects: bearer-, tele- and supplementary services, use of services, types and features of mobile stations (MS), licensing and subscription, as well as transferred and international accounting, etc.

R.03 - Network Aspects, including network functions and architecture, call routing to the MS, technical performance, availability and reliability objectives, handover and location registration procedures as well as discontinuous and cryptological algorithms, etc.

R.04 - Mobile / Base station (BS) interface and protocols, including specifications for layer 1 and 3 aspects of the open systems interconnection (OSI) seven-layer structure.

R.05 - Physical layer on the radio path, incorporating issues of multiplexing and multiple access, channel coding and modulation, transmission and reception, power control, frequency allocation and synchronisation aspects, etc.

R.06 - Speech coding specifications, such as functional, computational, and verification procedures for the speech codec and its associated voice activity detector (VAD) and other optional features.

R.07 - Terminal adaptor for MSs, including circuit and packet mode as well as voice-band data services.

R.08 - Base station (BS) and mobile switching centre (MSC) interface, and transcoder functions.

R.09 - Network interworking with the public telephone network (PSTN), integrated services digital network (ISDN) and packet data networks.

R.10 - Service internetworking, short message service.

R.11 - Equipment specification and type approval specification as regards to MSs, BSs, MSCs, home (HLR) and visited location register (VLR) as well as system simulator.

R.12 - Operation and maintenance, including subscriber, routing tariff and traffic administration as well as BS, MSC, HLR and VLR maintenance issues.

This treatise does not provide a full description of the GSM recommendations, our main intention is to offer a rudimentary exposure to the fundamental operation of the GSM system for the practitioner. We will mainly concentrate on series 05 and 06 Recommendations, dealing with the physical layer on the radio path as well as aspects of speech, data and signalling transmissions.

In Part I of the paper after a system overview in section 2 we embark upon describing aspects of mapping logical channels onto physical resources for speech, data and signalling channels in sections 4, 5 and 6, respectively. These details can be found in Recommendations R.05.02 and R.05.03. Synchronisation issues as well as frequency hopping (FH) [R.05.02.] and speech coding [R.06.10.] are considered in sections 7, 8 and 9 of Part I, respectively.

Modulation [R.05.04.], transmission via the standardised wide-band GSM channel models [R.05.05.] and Viterbi equalisation (VE) as well as adaptive radio link control [R.05.06.], [R.05.08.], discontinuous transmission (DTX) [R.06.31.], voice activity detection (VAD) [R.06.32.] and comfort noise insertion (CNI) along with ciphering issues [R.03.20.] are to be highlighted in Part II.

GLOSSARY

*A*3	Authentication algorithm
*A*5	Cyphering algorithm
*A*8	Confidential algorithm to compute the cyphering key
AB	Access burst
ACCH	Associated control channel
ADC	Administration centre
AGCH	Access grant control channel
AUC	Authentication centre
AWGN	Additive gaussian noise
BCCH	Broadcast control channel
BER	Bit error ratio
BFI	Bad frame indicator flag
BN	Bit number
BS	Base station
BS-PBGT	BS power budget: to be evaluated for power budget motivated handovers
BSIC	Base station identifier code
CC	Convolutional codec
CCCH	Common control channel
CELL-BAR-ACCESS	Boolean flag to indicate, whether the MS is permitted to access the specific traffic cell
CNC	Comfort noise computation
CNI	Comfort noise insertion
CNU	Comfort noise update state in the DTX handler
DB	Dummy burst
DL	Down link
DSI	Digital speech interpolation to improve link efficiency
DTX	Discontinuos transmission for power consumption and interference reduction
EIR	Equipment identity register
EOS	End of speech flag in the DTX handler
FACCH	Fast associated control channel
FCB	Frequency correction burst
FCCH	Frequency correction channel
FEC	Forward error correction
FH	Frequency hopping
FN	TDMA frame number
GMSK	Gaussian minimum shift keying
GP	Guard space
HGO	Hangover in the VAD
HLR	Home location register
HO	Handover
HOCT	Hangover counter in the VAD
HO-MARGIN	Handover margin to facilitate hysteresis
HSN	Hopping sequence number: frequency hopping algorithm's input variable
IMSI	International mobile subscriber identity
ISDN	Integrated services digital network
LAI	Location area identifier
LAR	Logarithmic area ratio
LTP	Long term predictor
MA	Mobile allocation: set of legitimate RF channels, input variable in the frequency hopping algorithm
MAI	Mobile allocation index: output variable of the FH algorithm
MAIO	Mobile allocation index offset: initial RF channel offset, input variable of the FH algorithm
MS	Mobile station
MSC	Mobile switching centre
MSRN	Mobile station roaming number
MS-TXPWR-MAX	Maximum permitted MS transmitted power on a specific traffic channel in a specific traffic cell
MS-TXPWR-MAX(n)	Maximum permitted MS transmitted power on a specific channel in the n-th adjacent traffic cell

NB	Normal burst
NMC	Network management centre
NUFR	Receiver noise update flag
NUFT	Noise update flag to ask for SID frame transmission
OMC	Operation and maintenance centre
PARCOR	Partial correlation
PCH	Paging channel
PCM	Pulse code modulation
PIN	Personal identity number for MSs
PLMN	Public land mobile network
PLMN-PERMITTED	Boolean flag to indicate, whether the MS is permitted to access the specific PLMN
PSTN	Public switched telephone network
QN	Quarter bit number
R	Random number in the authentication process
RA	Rural area channel impulse response
RACH	Random access channel
RF	Radio frequency
RFCH	Radio frequency channel
RFN	Reduced TDMA frame number: equivalent representation of the TDMA frame number, which is used in the synchronisation channel
RTNTABLE	Random number table utilised in the frequency hopping algorithm
RPE	Regular pulse excited
RPE-LTP	Regular pulse excited codec with long term predictor
RS-232	Serial data transmission standard equivalent to CCITT V24, interface
RXLEV	Received signal level: parameter used in handovers
RXQUAL	Received signal quality: parameter used in handovers
S	Signed response in the authentication process
SACCH	Slow associated control channel
SB	Synchronisation burst
SCH	Synchronisation channel
SCPC	Single channel per carrier
SDCCH	Stand-alone dedicated control channel
SE	Speech extrapolation
SID	Silence identifier
SIM	Subscriber identity module in MSs
SPRX	Speech received flag
SPTX	Speech transmit flag in the D TX handler
STP	Short term predictor
TA	Timing advance
TB	Tailing bits
TCH	Traffic channel
TCH/F	Full-rate traffic channel
TCH/F2.4	Full-rate 2.4 kbit/s data traffic channel
TCH/F4.8	Full-rate 4.8 kbit/s data traffic channel
TCH/F9.6	Full-rate 9.6 kbit/s data traffic channel
TCH/FS	Full-rate speech traffic channel
TCH/H	Half-rate traffic channel
TCH/F2.4	Half-rate 2.4 kbit/s data traffic channel
TCH/F4.8	Half-rate 4.8 kbit/s data traffic channel
TDMA	Time division multiple access
TMSI	Temporary mobile subscriber identifier
TN	Time slot number
TU	Typical urban channel impulse response
TXFL	Transmit flag in the DTX handler
UL	Up link
VAD	Voice activity detection
VE	Viterbi equaliser

2. OVERVIEW

The system elements of a GSM public land mobile network (PLMN) are portrayed in Fig. 1, where their interconnections via standardised interfaces are seen as well. The mobile station (MS) communicates with the serving and adjacent base stations (BS) via the radio interface U_m, while the BSs are connected to the mobile switching centre (MSC) through the network interface *A*. The BS is responsible for channel allocation [R.05.09.], link quality and power budget control [R.05.06.], [R.05.08.], signalling and broadcast traffic control, frequency hopping (FH) [R.05.02.], handover (HO) initiation [R.03.09.], [R.05.08.], etc. The MSC represents the gateway to other networks, such as the public switched telephone network (PSTN), integrated services digital network (ISDN) and packet data networks using the interworking functions standardised in [R.09.xx.]. The MSC's further functions include paging, MS location updating [R.03.12.], HO control [R.03.09], etc. The MS's mobility management is assisted by the home location register (HLR)[R.03.12], storing part of the MS's location information and routing incoming calls to the visitor location register (VLR) [R.03.12] in charge of the area, where the paged MS roams. Location update is asked for by the MS, whenever it detects from the received and decoded broadcast control channel (BCCH) messages that it entered a new location ar-

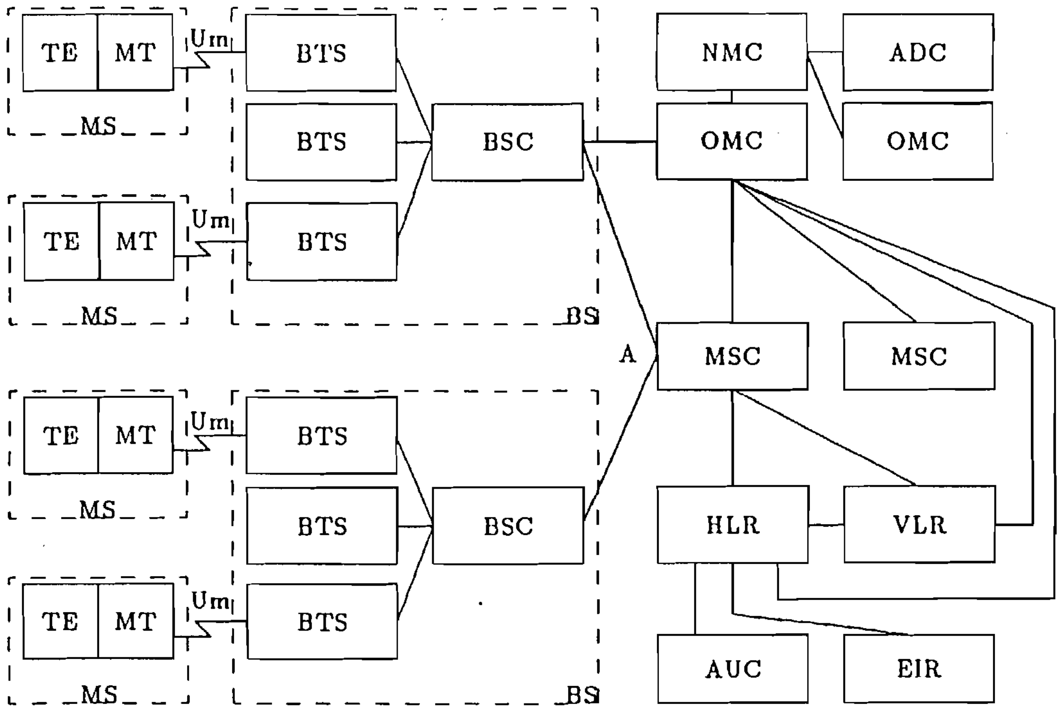


Fig. 1 - Simplified structure of GSM PLMN.

ea. The HLR contains, amongst a number of other parameters, the International Mobile Subscriber Identity (IMSI), which is used for the authentication [R.03.20] of the subscriber by his AUthentication Centre (AUC). This enables the system to confirm that the subscriber is allowed to access it. Every subscriber belongs to a home network and the specific services which the subscriber is allowed to use are entered into his HLR. The Equipment Identity Register (EIR) allows for stolen, fraudulent or faulty mobile stations to be identified by the network operators. The VLR is the functional unit that attends to a MS operating outside the area of its HLR. The visiting MS is automatically registered at the nearest MSC and the VLR is informed of the MSs arrival. A roaming number is then assigned to the MS and this enables calls to be routed to it. The Operations and Maintenance Centre (OMC), Network Management Centre (NMC)and ADministration Centre (ADC) are the functional entities through which the system is monitored, controlled, maintained and managed [R.12.xx.].

The MS initiates a call by searching for a BS with a sufficiently high received signal level on the BCCH carrier, it will await and recognise a frequency correction burst and synchronise to it [R.05.08]. Now the BS allocates a bidirectional signalling channel and also sets up a link with the MSC via the network. The MSC uses the IMSI received from the MS to interrogate its HLR and sends the data obtained to the serving VLR. After authentication [R.03.20] the MS provides the destination number, the BS allocates a traffic channel and the MSC routes the call to its destination. If the MS moves to another cell, it is re-assigned to another BS and a handover occurs. If both BSs in the handover process are controlled by the same BSC the handover takes place under the control of the BSC, otherwise it is performed by the MSC. In case of incoming calls the MS must be paged by the BSC. A paging signal is transmitted on a paging channel (PCH) monitored continuously by all MSs and covers the location area in which the MS roams. In response to the paging signal the MS performs an access procedure identical to that employed when the MS initiates a call.

3. LOGICAL AND PHYSICAL CHANNELS

The GSM logical traffic and control channels are standardised in Recommendation [R.05.02], while their mapping onto physical channels is the subject of [R.05.02] and [R.05.03]. The GSM system's prime objective is to transmit the logical traffic channel's (TCH) speech or data information. Their transmission via the network requires a variety of logical control channels. The set of logical traffic and control channels defined in the GSM system is summarised in Table 2. There are two general forms of speech and data traffic channels: the full rate traffic channels (TCH/F), which carry information at a gross rate of 22.8 kbit/s, and the half rate traffic channels (TCH/H), which communicate at a gross rate of 11.4 kbit/s. A physical channel carries either a full rate traffic channel, or two half rate traffic channels. In the former the traffic channel occupies one timeslot, while in the latter the two half-rate traffic channels are mapped onto the same timeslot, but in alternate frames.

For a summary of the logical control channels carry-

Table 2 - *GSM Logical Channels*

Logical Channels					
Duplex BS - MS Traffic Channels: TCH		Control Channels: CCH			
FEC-coded Speech TCH/F 22.8 kbit/s	FEC-coded Data TCH/F9.6 TCH/F4.8 TCH/F2.4 22.8 kbit/s	Broadcast CCH BCCH BS $\rightarrow$ MS	Common CCH CCCH	Stand-alone Dedicated CCH SDCCH BS $\rightarrow$ MS	Associated CCH ACCH BS $\rightarrow$ MS
		Freq. Corr. Ch: FCCH	Paging Ch: PCH BS $\rightarrow$ MS	SDCCH/4	Fast ACCH: FACCH/F, FACCH/H
TCH/H 11.4 kbit/s	TCH/H4.8 TCH/H2.4 11.4 kbit/s	Synchron. Ch: SCH	Random Access Ch: RACH MS $\rightarrow$ BS	SDCCH/8	Slow ACCH: SACCH/TF, SACCH/TH SACCH/C4, SACCH/C8
		General Inf.	Access Grant Ch: AGCH BS $\rightarrow$ MS		

ing signalling or synchronisation data see Table 2. There are four categories of logical control channels, known as the broadcast control channel (BCCH), the common control channel (CCCH), the stand-alone dedicated control channel (SDCCH) and the associated control channel (ACCH). The purpose and way of deployment of the logical traffic and control channels will be explained by highlighting, how they are mapped onto physical channels in assisting high-integrity communications.

A physical channel in a TDMA system is defined as a timeslot with a timeslot number (TN) in a sequence of TDMA frames. However, the GSM system deploys TDMA combined with frequency hopping and hence the physical channel is partitioned in both time and frequency. Consequently the physical channel is defined as a sequence of radio frequency channels and timeslots. Each carrier frequency supports 8 physical channels mapped onto 8 timeslots within a TDMA frame. A given physical channel always uses the same timeslot number TN in every TDMA frame. Therefore, a timeslot sequence is defined by a timeslot number TN and a TDMA frame number (FN) sequence.

4. SPEECH TRANSMISSION

The speech coding standard is [R.06.10.], while issues of mapping the logical speech traffic channel's information onto the physical channel constituted by a timeslot of a certain carrier are specified in [R.05.02.]. Since the error correction coding constitutes part of this mapping process, also [R.05.03.] is relevant to these discussions. The example of the full rate speech traffic channel (TCH/FS) is used to highlight, how this logical channel is mapped onto the physical channel constituted by a so-called Normal Burst (NB) of the TDMA frame structure. This mapping is explained by referring to Fig. 2 and Fig. 3. Then this example will be extended to other physical bursts such as the Frequency Correction (FCB), Synchronisation (SB), Access (AB) and Dummy Burst (DB) carrying logical control channels, as well as to their TDMA frame structures, as seen in Fig. 2 and Fig. 8.

The Regular Pulse Excited (RPE) speech encoder delivers 260 bits/20 ms at a bitrate of 13 kbit/s, which are devided into three significance classes: Class 1a (50 bits), Class 1b (132 bits) and Class 2 (78 bits). The Class 1a bits are encoded by a systematic (53, 50) cyclic error detection code by adding three parity bits. Then the bits are reordered and four zero tailing bits are added to perodically reset the subsequent half rate, constraint length five convolutional codec (CC) CC (2, 1, 5), as portrayed in Fig. 3. Now the unprotected 78 Class 2 bits are concatenated to yield a block of 456 bits/20 ms, which implies an encoded bitrate of 22.8 kbit/s. This frame is partitioned into eight 57 bit subblocks that are blockdiagonally interleaved before undergoing intraburst interleaving. At this stage each 57 bit subblock is combined with a similar subblock of the previous 456 bit frame to construct a 116 bit burst, where the flag bits *hl* and *hu* are included to classify, whether the burst being transmitted is really a TCH/FS burst or it has been 'stolen' by an urgent (FACCH) message. Now the bits are encrypted and positioned in a Normal Burst (NB), as depicted at the bottom of Fig. 2, where three tailing bits (TB) are added at both ends of the burst to reset the memory of the Viterbi equaliser (VE). This mapping process is also summarised in the blockdiagram of Fig. 4.

The 8.25 bit-interval duration guard space (GP) at the bottom of Fig. 2 is provided to prevent burst overlapping due to delay fluctuations. Finally, a 26 bit equaliser training segment is included in the centre of the normal traffic burst. This segment is constructed by a 16 bit Viterbi channel equaliser training pattern surrounded by five quasi-periodically repeated bits on both sides. Since the MS has to be informed about which BS it communicates with, for neighbouring BSs one of eight different training patterns is used, associated with the BS colour codes.

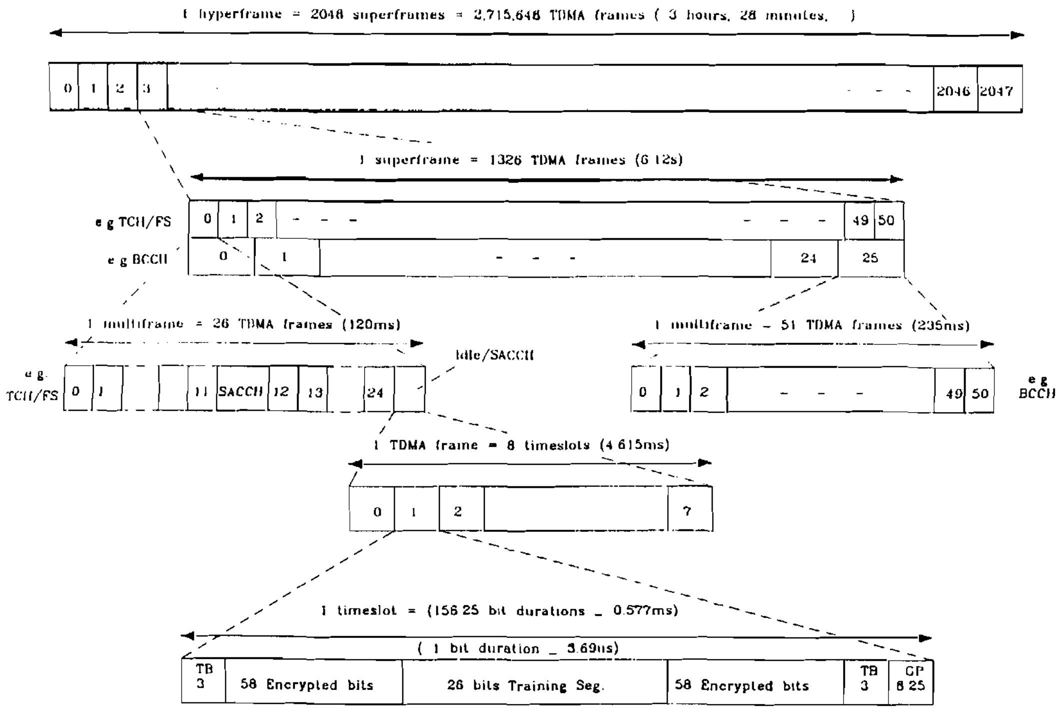


Fig. 2 - The GSM TDMA frame structure.

This 156.25 bit duration TCH/FS normal burst (NB) constitutes the basic timeslot of the TDMA frame structure, which is input to the Gaussian minimum shift keying (GMSK) modulator to be highlighted in Part II at a bitrate of approximately 271 kbit/s. Since the bit interval is 3.69 μs, the timeslot duration is $156.25 \cdot 3.69 \approx 0.577$ ms. Eight such normal bursts of eight appropriately staggered users are multiplexed onto one (RF) carrier giving, a TDMA frame of ms duration, as we see in Fig. 2. The physical channel as characterised above provides a physical timeslot with a throughput of 114 bits/4.615 ms = 24.7 kbit/s, which is sufficiently high to transmit the 22.8 kbit/s TCH/FS. It even has a 'reserved' capacity of 24.7-22.8 = 1.9 kbit/s, which can be exploited to transmit slow control information associated with this specific traffic channel, i.e., to construct a so-called Slow Associated Control Channel (SACCH), constituted by the SACCH TDMA frames, interspersed with traffic frames at multiframe level of the hierarchy, as seen in Fig. 2

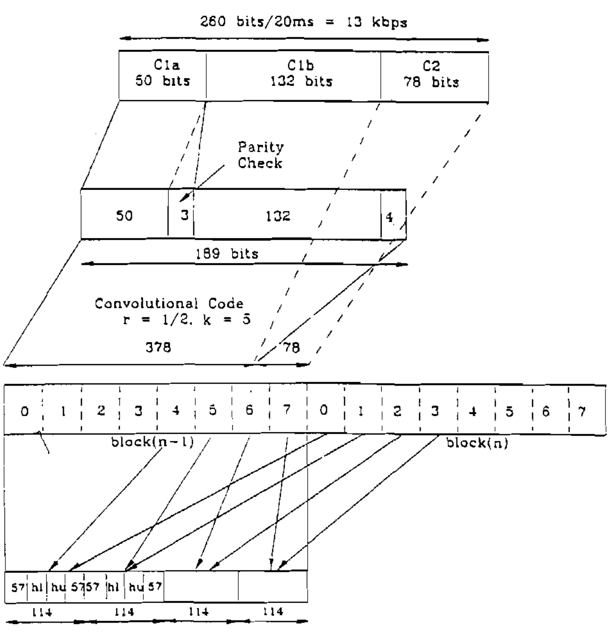


Fig. 3 - Mapping the TCH/FS logical channel onto a physical channel.

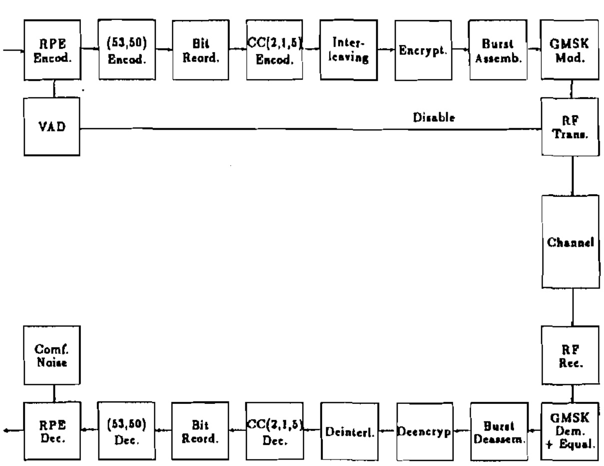


Fig. 4 - Blockdiagram of the TCH/FS channel.

5. DATA TRANSMISSION

Mapping logical data traffic channels onto a physical channel is essentially carried out by the channel codecs specified in [R.05.03.].The full-and half-rate data traffic channels standardised in the GSM system are: *TCH / F*9.6, *TCH / F*4.8, *TCH / F*2.4, as well as *TCH/H*4.8, *TCH/H*2.4, but here we only consider the mapping of *TCH / F*9.6. The data user unit's interface defined in R.04.21 delivers blocks of 60 bits every 5ms, out of which 48 bits are useful data and 12 are reserved for example for *RS*-232 standard signalling. Clearly, 9.6 kbit/s useful data is transmitted via a 12kbit/s link.The burst structure of *TCH / F*9.6 is explained by referring to Fig. 5. Four consecutive 60 bit data blocks are arranged to constitute a 240 bit information block, which is followed by four zero tailing bits for resetting the subsequent punctured convolutional codec after the transmission of a frame. The 240 bit data blocks are encoded by the half-rate, constraint length $K = 5$ punctured convolutional code PCC(2,1,5) using the same generator polynomials $G_0 = 1 + D^3 + D^4$ and $G_1 = 1 + D + D^3 + D^4$, as in the speech channel. This *PCC*(2, 1, 5) convolutional code produces 488 encoded bits from the 244 data bits, but the following 32 coded bits are consistently punctured, i.e., not transmitted:$c(11 + 15j)$; $j = 0, 1, \ldots 31$. The 456 encoded bits $c(0) \ldots c(455)$ are then mapped onto four consecutive 114-bit TDMA bursts (K, $K+1$, $K+2$ and $K+3$) using the mapping rule: $b(K, k) = c(k)$, $b(K+1, k) = c(k + 144)$, $b(K+2, k) = c(k + 228)$, $b(K+3, k) = c(k + 342)$, where $k = 0, 1 \ldots 113$.

The stealing flags *hl* and *hu* used in the full-rate speech channel TCH/FS are included also here in the centres of the bursts, as seen in Fig. 5b) and have the same interpretation as in TCH/FS. The encoded bits are now reordered according to the following inter-burst interleaving rule: $i(K, j) = c(n, k)$, $K = K_0 + n + [k \bmod 19]$, $j = [k \bmod 19] + 19[k \bmod 6]$, $k = 0 \ldots 113$, where k is the bit-index in the 114 bit encoded bursts, n is the encoded burst-index, K is the interleaved burst-index and j is the bit-index in the interleaved burst. Inter-burst interleaving is known to possess good randomising properties in dispersing bursty channel errors, particularly, if the interleaving memory is sufficiently long. In this scheme $N \cdot B = 6 \cdot 19 = 114$ bits of an encoded burst are dispersed over $N = 19$ consecutive interleaved bursts, including the current one, while donating $B = 6$ bits to each one of them. A representative example of mapping the encoded bits of the K-th encoded burst $c(K, k)$, $k = 0 \ldots 114$ onto bits of the subsequent 19 interleaved bursts is given in Fig. 5c). Viewing the mapping from a different angle, bits 0, 19, 38, 57 etc. of the K-th encoded burst are mapped onto itself, then bits 1, 20, 29, 58 etc. onto burst ($K + 1$), etc. This arrangement disperses 114 bits of a burst over $19 \cdot 114 = 2166$ bits, which appears sufficiently long, when combined with frequency hopping to randomise the bursty error statistics even for the slowly fading received signal envelopes of pedestrians as well.

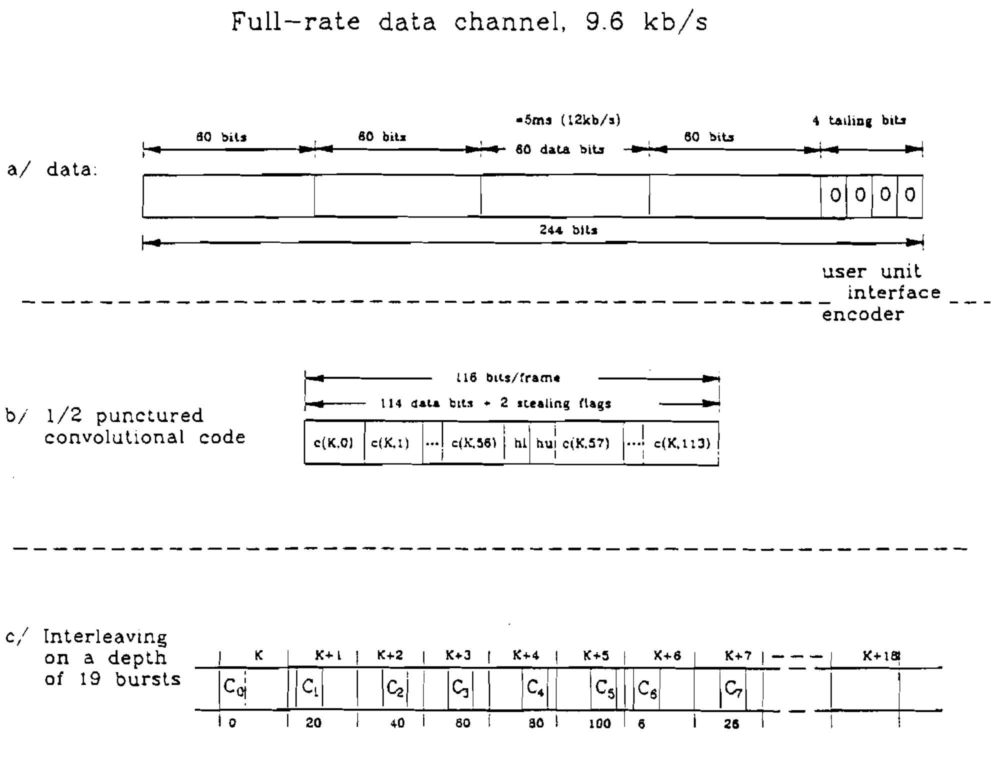


Fig. 5 - TCH/F9.6 FEC and burst structure.

6. SIGNALLING TRANSMISSION

The exact derivation, FEC coding and mapping of logical control channel information is beyond the scope of this paper, the interested reader is referred to [R.05.02.], [R.05.03.] and [1] for a detailed discussion. As an example, the mapping of the 184 bit slow associated control channel (SACCH), fast associated control channel (FACCH), broadcast control channel (BCCH), standalone dedicated control channel (SDCCH), paging channel (PCH) and access grant control channel (AGCH) messages onto a 456 bit block, i.e. onto four 114-bit bursts is demonstrated in Fig. 6. A double-layer concatenated FIRE-code/convolutional code scheme generates 456 bits, using an overall coding-rate of R = 184/456, which gives a stronger protection for control channels than the error protection of traffic channels.

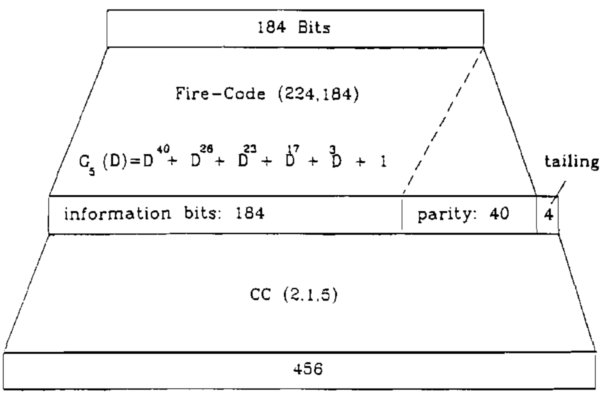


Fig. 6 - FEC in SACCH, FACCH, BCCH, SDCCH, PCH and AGCH.

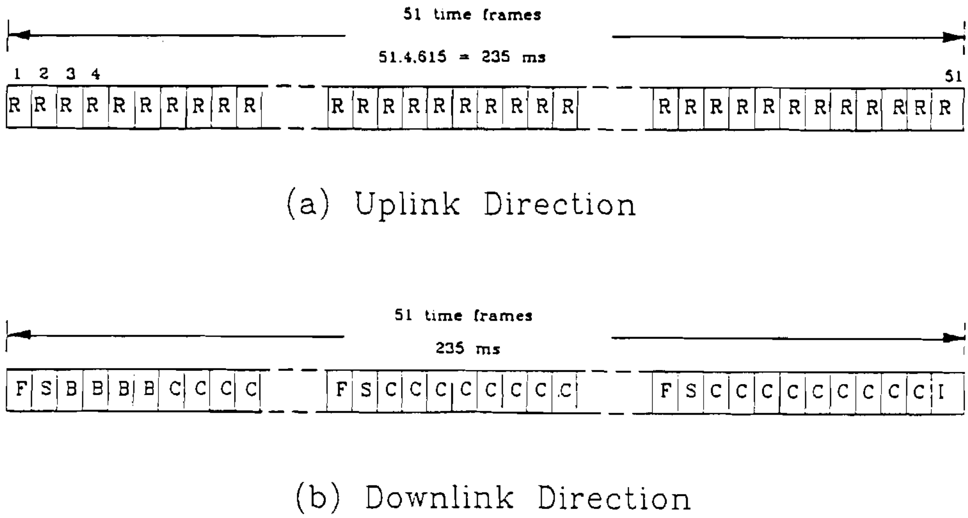
The SACCH is accommodated by the TDMA frame-structure in the following fashion. The TCH/FS TDMA frames of the eight users are multiplexed into multiframes of 24 TDMA frames, but the 13-th frame will carry a SACCH message, rather than the 13-th TCH/FS frame, while the 26-th frame will be an idle or dummy frame, as seen at the left hand side of Fig. 2 at the multiframe level of the traffic channel hierarchy. The general control channel frame structure shown at the right of Fig. 2 is discussed later. This way 24 TCH/FS frames are sent in a 26-frame multiframe during 26 · 4.615 = 120 ms. This reduces the traffic throughput to 24/26 · 24.7 = 22.8 kbit/s required by TCH/FS, allocates 1/26 · 24.7 = 950 bit/s to the SACCH and 'wastes' 950 bit/s in the idle frame. Observe that the SACCH frame has eight timeslots to transmit the eight 950 bit/s SACCHs of the eight users on the same carrier. The 950 bit/s idle capacity will be used in case of half rate channels, where 16 users will be multiplexed onto alternate frames of the TDMA structure to increase system capacity, when a half rate speech codec becomes available. Then sixteen 11.4 kbit/s encoded TCH/HSs will be transmitted in a 120 ms multiframe, where also sixteen SACCHs are available.

The Fast Associated Control Channel (FACCH) messages are transmitted via the physical channels provided by bits 'stolen' from their own host traffic channels. The construction of the FACCH bursts from 184 control bits is identical to that of the SACCH, as also shown in Fig. 6, but its 456 bit frame is mapped onto eight consecutive 114 bit TDMA traffic bursts, exactly, as specified for TCH/FS. This is carried out by stealing the even bits of the first four and the odd bits of the last four bursts, which is signalled by setting $hu = 1$, $hl = 0$ and $hu = 0$, $hl = 1$ in the first and last bursts, respectively. The unprotected FACCH information rate is 184 bits/20 ms = 9.2 kbit/s, which is transmitted after concatenated error protection at a rate of 22.8 kbit/s. The repetition delay is 20 ms and the interleaving delay is 8 · 4.165 = 37 ms, resulting in a total of 57 ms delay.

At the next hierarchical level of Fig. 2 51 TCH/FS multiframes are multiplexed into one superframe lasting 51 · 120 ms = 6.12 s, which contains 26 · 51 = 1326 TDMA frames. However, with FN = 1326 number of TDMA frames the encryption rule relying on this value is not sufficiently secure. Whence 2048 superframes form a hyperframe of 1326 · 2048 = 2715648 TDMA frames lasting 2048 · 6.12 s ≈ 3 h 28 min, using a sufficiently high FN in the encryption algorithm. The uplink and downlink traffic-frame structures are identical with a shift of three timeslots between them, which relieves the MS from having to transmit and receive simultaneously and also allows monitoring the powers of adjacent BSs in unallocated timeslots.

In contrast to duplex traffic and associated control channels, the simplex BCCH and CCCH logical channels of all MSs roaming in a specific cell share the physical channel provided by timeslot zero of the so-called BCCH carriers available in the cell. Furthermore, 51 BCCH and CCCH TDMA frames are mapped onto a 51 · 4.615 = 235 ms duration multiframe, rather than on a 26-frame, 120 ms duration multiframe. To compensate for the extended multiframe length of 235 ms, 26 multiframes constitute a 1326-frame superframe of 6.12 s duration, as demonstrated by Fig. 2. Also the allocation of the uplink and downlink frames is different, since these control channels exist only in one direction, as seen in Fig. 7.

The random access channel (RACH) is only used by the MSs in uplink direction if they request, for example, a bidirectional stand-alone dedicated control channel (SDCCH) to be mapped onto an RF channel to register with the network and set up a call. The uplink RACH carries messages of eight bits per 235 ms multiframe, which is equivalent to an unprotected control information rate of 34 bit/s. These messages are concatenated FEC coded to a rate of 36 bits/235 ms = 153 bit/s. They are not transmitted by the Normal Bursts (NB) derived for TCH/FS, SACCH or FACCH logical channels, but by the so-called Access Bursts (AB), depicted in Fig. 8 in comparison to a NB and other types of bursts to be



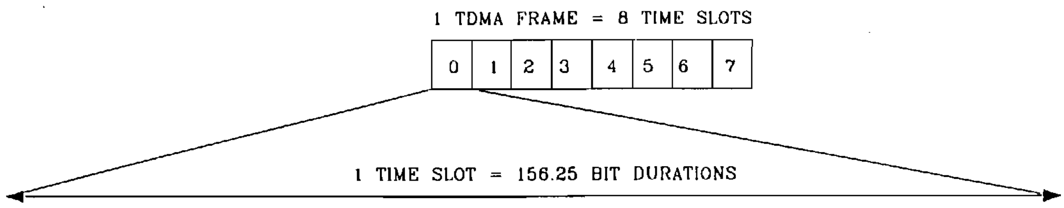
R: Random Access Channel
F: Frequency Correction Channel
S: Synchronisation Channel
B: Broadcast Control Channel
C: Access Grant/Paging Channel
I: Idle Frame

Fig. 7 - The control multiframe.

described later. The FEC coded, encrypted 36 bit messages, containing amongst other parameters also the encoded 6 bit BS identifier code (BSIC) constituted by the 3 bit PLMN colour code and 3 bit BS colour code for unique BS identification, are positioned after the 41 bit synchronisation sequence, which is extended to ensure reliable access burst recognition. These messages have no interleaving delay, while they are transmitted with a repetition delay of one control multiframe length, i.e. 235 ms.

Adaptive time frame alignment is a technique designed to equal out propagation delay differences between MSs at different distances. The GSM system is designed to allow cell sizes up to 35 km radius. The time a radio signal takes to travel the 70 km from the base station to the mobile station and back again is 233.3 μs. As signals from all the mobiles in the cell must reach the base station without overlapping each other, a guard period of 68.25 bits (252 μs) is provided in the access burst. This long guard period in the access burst is needed when the mobile station attempts its first access to the base station, or after handover has occurred. When the base station detects a 41 bit random access synchronisation sequence with a long guard period, it measures the received signal delay relative to the expected signal from a mobile station of zero range. This delay, called the timing advance, is signalled using a 6 bit number to the mobile station, which advances its timebase over the range of 0 to 63 bits, i.e., in units of 3.69 μs. By this process the TDMA bursts arrive at the BS in their correct timeslots and do not overlap with adjacent ones. This process allows the guard period in all other bursts to be reduced to $8.25 \cdot 3.69\ \mu s \approx 30.46\ \mu s$ (8.25 bits) only. In normal operation the BS continously monitors the signal delay from the MS and will instruct the MS to update its time advance parameter. In very large traffic cells there is an option to actively utilise every second timeslot only to cope with higher propagation delays, which is spectrally inefficient, but in these large, low-traffic rural cells admissible.

The downlink multiframe transmitted by the BS is shared amongst a number of BCCH and CCCH logical channels, as depicted in Fig. 7. In particular, the last frame is an idle frame (I), while the remaining 50 frames are divided in five blocks of ten frames, where each block starts with a frequency correction channel (FCCH) followed by a synchronisation channel (SCH). In the first block of ten frames the FCH and SCH



NORMAL BURST

TAIL BITS	ENCRYPTED BITS	TRAINING SEQUENCE	ENCRYPTED BITS	TAIL BITS	GUARD PERIOD
3	58	26	58	3	8.25

FREQUENCY CORRECTION BURST

TAIL BITS	FIXED BITS	TAIL BITS	GUARD PERIOD
3	142	3	8.25

SYNCHRONISATION BURST

TAIL BITS	ENCRYPTED SYNC BITS	EXTENDED TRAINING SEQUENCE	ENCRYPTED SYNC BITS	TAIL BITS	GUARD PERIOD
3	39	64	39	3	8.25

ACCESS BURST

TAIL BITS	SYNCHRO SEQUENCE	ENCRYPTED BITS	TAIL BITS	GUARD PERIOD
8	41	36	3	68.25

Fig. 8 - GSM burst structures.

frames are followed by four broadcast control channel (BCCH) frames and by either four access grant control channels (AGCH) or four paging channels (PCH). In the remaining four blocks of ten frames the last eight frames are devoted to either PCHs or AGCHs, which are mutually exclusive for a specific MS being either paged or granted a control channel.

The frequency correction channel (FCCH), synchronisation channel (SCH) and random access channel (RACH) require special transmission bursts, tailored to their missions, as depicted in Fig. 8. Namely, the FCCH uses frequency correction bursts (FCB) with a fixed 142 bit pattern, which result in a modulated signal represented by an unmodulated carrier, exhibiting a fixed frequency offset from the RF carrier utilised. The synchronisation channel deploys synchronisation bursts (SB) with a $16 \cdot 4 = 64$ bit extended training sequence to allow frame alignment with a quarter bit accuracy, as well as the transmission of BS and PLMN colour codes, representing one of eight legitimate identifiers in the encrypted synchronisation bits. Lastly, the access bursts (AB) with an extended 41 bit synchronisation sequence are invoked to facilitate initial access to the system with a long guard space of 68.25 bit duration, before the MS's distance, i.e. propagation delay becomes known to the BS and it can be compensated for by adjusting the MS's timing advance.

7. SYNCHRONISATION ISSUES

Although some synchronisation issues are standardised in [R.05.02.], [R.05.03.], the GSM Recommendations do not specify the exact BS-MS synchronisation algorithms to be used, these are left to the equipment manufacturers. However, a unique set of timebase counters is defined to ensure perfect BS-MS synchronism. The BS sends frequency correction bursts (FCB) and synchronisation bursts (SB) on specific timeslots of the BCCH carrier to the MS to ensure that the MS's frequency standard is perfectly aligned with that of the BS, as well as to inform the MS about the required initial state of its internal counters. The MS sends its uniquely numbered traffic and control bursts staggered by three timeslots with respect to those of the BS to prevent simultaneous MS transmission and reception, and also takes into account the required timing advance (TA) to cater for different BS-MS-BS round-trip delays.

The timebase counters used to uniquely describe the internal timing states of BSs and MSs are the Quarter bit Number (QN = 0 … 624) counting the quarter bit intervals in bursts, Bit Number (BN = 0 … 156), Timeslot Number (TN = 0 … 7) and TDMA Frame Number (FN = $0 \ldots 16 \cdot 51 \cdot 2048$), given in the order of increasing interval duration. The MS sets up its timebase counters after receiving a SB by determining QN from the 64 bit extended training sequence in the centre of the SB, setting TN = 0 and decoding the 78 encrypted, protected bits carrying the 25 SCH control bits.

The SCH carries frame synchronisation information as well as BS identification information to the MS, as seen in Fig. 9, and it is provided solely to support the operation of the radio subsystem. The first six bits of

PLMN colour 3 bits	BS colour 3 bits	T1:superframe index 11 bits	T2:multiframe index 5 bits	T3:block frame index 3 bits
BSIC 6 bits		RFN 19 bits		

Fig. 9 - Synchronisation channel (SCH) message format.

the 25 bit segment consist of three PLMN colour code bits and three BS colour code bits supplying a unique BS Identifier Code (BSIC) to inform the MS, which BS it is communicating with. The second 19 bit segment is the so-called Reduced TDMA Frame Number (RFN) derived from the full TDMA Frame Number (FN), constrained to the range of $[0 \ldots (26 \cdot 51 \cdot 2048) - 1] = [0 \ldots 2,715,647]$ in terms of three subsegments $T1$, $T2$ and $T3$. These subsegments are computed as: $T1$ (11 bits) = $[FN \text{ div } (26 \cdot 51)]$, $T2$ (5 bits) = $[FN \bmod 26]$ and $T3'$ (3 bits) = $[(T3 - 1) \text{ div } 10]$, where $T3 = [FN \bmod 5]$. Here $T1$ determines the superframe index in a hyperframe, $T2$ the multiframe index in a superframe, $T3$ the frame index in a multiframe, while $T3'$ the block index of a frame in a specific control multiframe, where their role is best understood by referring to Fig. 2. Once the MS has received the Synchronisation Burst (SB), it readily computes the FN required in various control algorithms, such as encryption, handover, etc., as shown below:

$$FN = 51\left[(T3 - T2) \bmod 26\right] + T3 + 51 \cdot 26 \cdot T1,$$
$$\text{where } \; T3 = 10 \cdot T3' + 1$$

It is desirable to have perfect synchronism of all the channels under the control of a BS, and hence all its RF carrier frequencies and timebase counter frequencies are derived from the same reference frequency of 13 MHz. It is possible but not mandatory to synchronise different BSs together. When the BS detects the AB of a RACH message with its 41 bit synchronisation sequence, its unique BSIC and 68.25 bit guard period, it will notice that the MS is asking for random access to it. Using the 41 bit synchronisation sequence in this decoded AB the BS can now evaluate the propagation delay, which will be the timing advance to be signalled, rounded to the nearest integer bit period, to the MS. As the timing advance is encoded by 6 bits, it is hard-limited to $64 \cdot 3.69 = 236\ \mu s$ and it is kept constant for higher propagation delays. This timing advance is continuously updated and the adjustment error is less than half of a bit period.

8. FREQUENCY HOPPING

Frequency hopping [R.05.02.] combined with interleaving is known to be very efficient in combating channel fading, and it results in near-Gaussian performance even over hostile Rayleigh-fading channels. The principle of Frequency Hopping (FH) is that each TDMA burst is transmitted via a different RF CHannel (RFCH). If the present TDMA burst happened to be in a deep fade, then the next burst most probably will not be, as long as hopping is carried out to a frequency outside the coherence bandwidth $B_c = 1/2\ \pi d$ of the dispersive channel, where d is the channel's delay-spread. In this sense GSM operators are likely to benefit from being assigned carriers at the extreme edges of the GSM-band, rather than being confined to a narrow block of frequencies. Either way, FH reduces the amount of time spent by the MS in a fade to 4.615 ms, the duration of a TDMA burst, which brings substantial gains in case of slowly moving MSs, such as pedestrians.

The GSM frequency hopping algorithm is shown in Fig. 10. The algorithm's input parameters include the TDMA Frame Number (FN) specified in terms of the indices $FN(T1)$, $FN(T2)$ and $FN(T3)$, as received in the synchronisation burst (SB) via the Synchronisation Channel (SCH). A further parameter is the set of RF channels called mobile allocation (MA) assigned for use in the MS hopping sequence, which is limited to $1 \le N \le 64$ channels out of the legitimate 124 GSM channels, but it is likely to be less than 32 due to a number of operators competing for frequencies. The Mobile Allocation Index Offset (MAIO) determines the minimum value of the Mobile Allocation Index (MAI), which is the output variable of the FH algorithm determining the next RF channel to which frequency hopping is required.

The Hopping Sequence generator Number (HSN) $0 \le \text{HSN} \le 63$ is a further control parameter, which results in cyclic hopping if HSN = 0, as seen in Fig. 10, and in pseudo-random hopping patterns if $1 \le \text{HSN} \le 63$. This is, because for HSN = 0 the mobile allocation index is computed as:

$$\text{MAI} = \left[(FN + \text{MAIO}) \bmod \; N\right]$$

where (mod N) is taken to ensure that MAI remains an element of the set MA. For $1 \le \text{HSN} \le 63$ somewhat more complex operations have to be computed, using a number of intermediate internal variables, as demonstrated by Fig. 10. The only undefined variable in the figure is NB, representing the number of bits required for the binary encoding of N, the number of RF channels in the set MA. The function RNTABLE simply assigns one out of 114 pseudo-random numbers specified by GSM according to its argument, the XOR operator

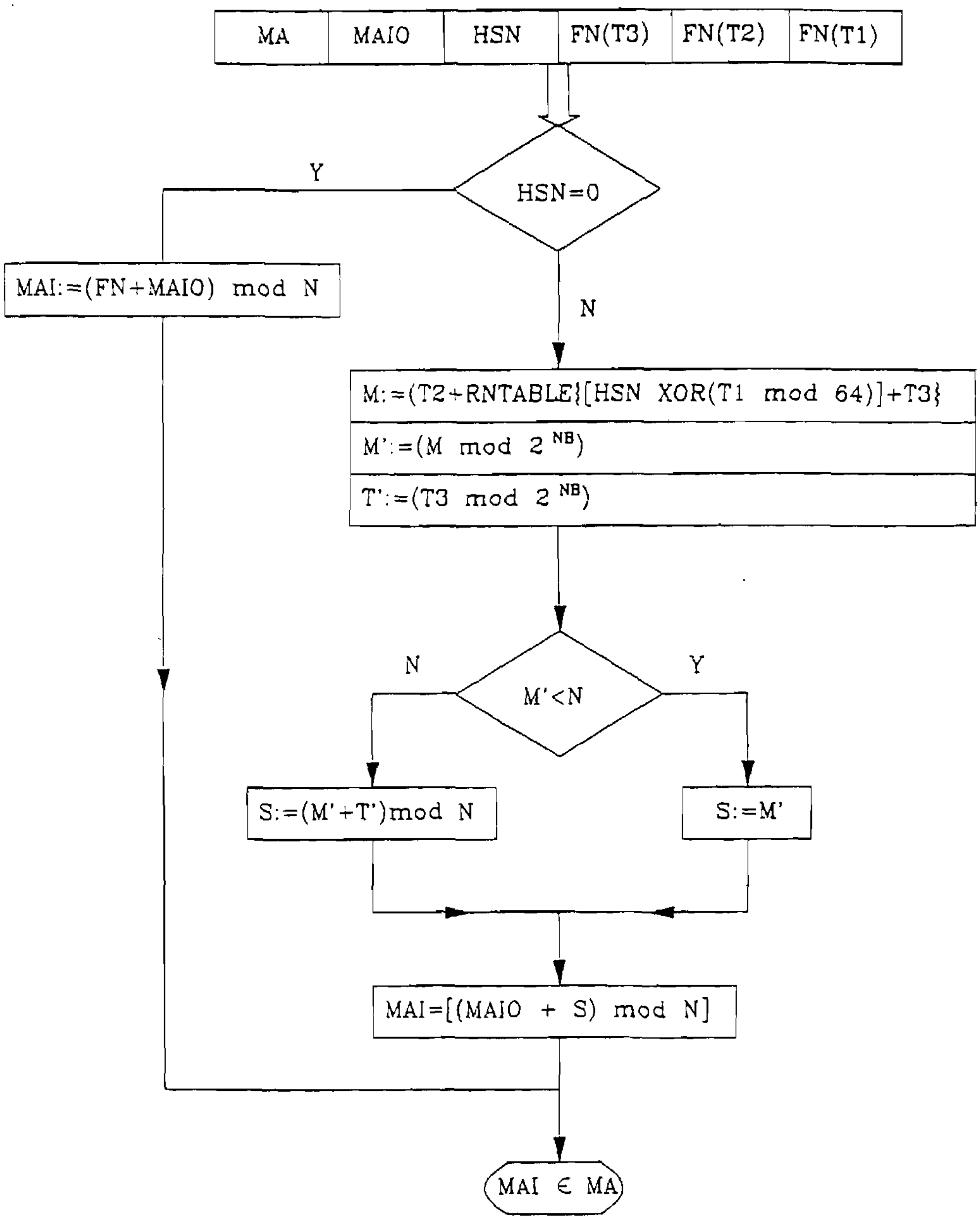


Fig. 10 - The GSM frequency hopping algorithm.

means bit-wise exclusive OR, while the remaining operations are self-explanatory. The outcome of the process is the mobile allocation index (MAI) specifying the next RF channel to be used by the MS. Note that frequency hopping is not allowed on timeslot zero of the BCCH carrier, which is ensured by using a single RFCH in the MA, i.e., setting $N = 1$ and MAIO = 0. In this case the FH sequence generation is uneffected by the value of HSN.

9. SPEECH CODING

The GSM speech coding algorithms to be used are stipulated in [R.06.10.]. The selection of the most appropriate speech codec for the GSM system from the set of candidate codecs was based on extensive comparative tests among various operating conditions. The rigorous comparisons published in [9] are interesting and offer deep insights for system designers as regards to the pertinent trade-offs in terms of speech quality, robustness against channel errors, complexity, system delay, etc. Here, however, we restrict our discussion to the standardised regular pulse excited (RPE) codec with short term (STP) and long term predictor (LTP).

The schematic diagram of the RPE-LTP encoder is shown in Fig. 11, where the following functional parts can be recognised [10 - 14]: 1. Pre-processing, 2. STP analysis filtering, 3. LTP analysis filtering, 4. RPE computation.

1 Pre-processing: Pre-emphasis can be deployed to increase the numerical precision in computations by emphasizing the high-frequency, low-power part of the speech spectrum. This can be carried out by the help of

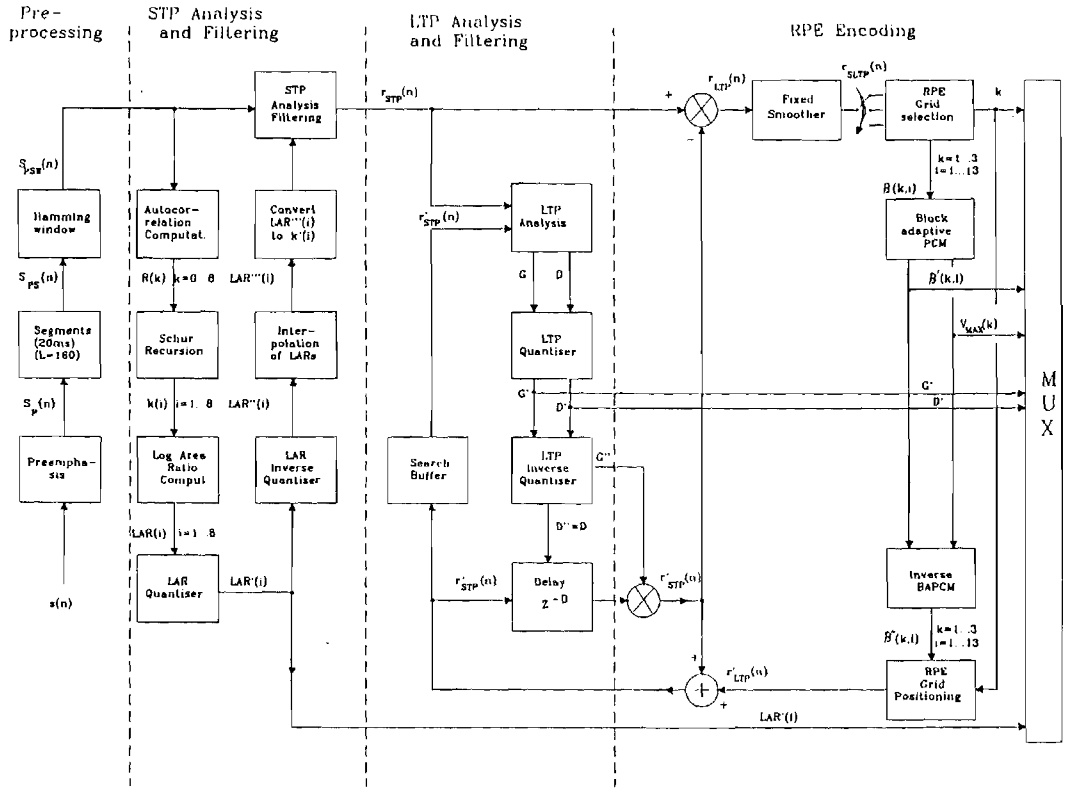


Fig. 11 - Blockdiagram of the RPE-LTP Encoder.

a one-pole filter with the transfer function of: $H(z) = 1 - c_1 z^{-1}$, where $c_1 \approx 0.9$ is a practical value. The pre-emphasized speech $s_p(n)$ is segmented into blocks of 160 samples in a buffer, where they are windowed by a Hamming-window to counteract the spectral domain Gibbs oscillation, caused by truncating the speech signal outside the analysis frame. The Hamming-window has a tapering effect towards the edges of a block, while it has no influence in its middle ranges: $s_{psw}(n) = s_{ps}(n) \cdot c_2 \cdot (0.54 - 0.46 \cos 2\pi n / L)$, where $s_{ps}(n)$ represents the pre-emphasized, segmented speech, $s_{psw}(n)$ is its windowed version and the constant $c_2 = 1.5863$ is determined from the condition that the windowed speech must have the same power as the non-windowed.

2 STP analysis filtering: For each segment of $L = 160$ samples nine autocorrelation coefficients $R(k)$ are computed from $s_{psw}(n)$ by: $R(k) = \sum_{n=0}^{L-1-k} s_{psw}(k) s_{psw}(n+k)$, $k = 0 \ldots 8$. From the speech autocorrelation coefficients $R(k)$ eight reflection coefficients k_i are computed according to the Schur-recursion [13], which is an equivalent method to the Durbin algorithm used for solving the LPC key-equations to derive the reflection coefficients k_i, as well as the STP filter coefficients a_i. However, the Schur-recursion delivers the reflection coefficients k_i only. The reflection coefficients k_i are converted to logarithmic area ratios (LAR(i)), because the logarithmically companded LARs have better quantisation properties than the coefficients k_i:

$$LAR(i) = \log_{10}\left(\frac{1+k(i)}{1-k(i)}\right)$$

where a piecewise linear approximation with five segments is used to simplify the real-time implementation. The various LAR(i) $i = 1 \ldots 8$ filter parameters have different dynamic ranges and differently shaped probability density functions (PDFs). This justifies the allocation of 6, 5, 4 and 3 bits to the first, second, third and fourth pairs of LARs, respectively. The quantised LAR(i) coefficients LAR′(i) are locally decoded into the set LAR″(i) as well as transmitted to the speech decoder. So as to mitigate the abrupt changes in the nature of the speech signal envelope around the STP analysis frame edges, the LAR parameters are linearly interpolated, and towards the edges of an analysis frame the interpolated LAR‴(i) parameters are used. Now the locally decoded reflection coefficients $k'(i)$ are computed by converting LAR‴(i) back into $k'(i)$, which are used to compute the STP residual $r_{STP}(n)$ in a PARCOR structure. The PARCOR scheme directly uses the reflection coefficients $k(i)$ to compute the STP residual $r_{STP}(n)$, and it constitutes the natural analogy to the acoustic tube model of the human speech production.

3 LTP analysis filtering: The LTP prediction error is minimised by that LTP delay D, which maximises the crosscorrelation between the current STP residual $r_{STP}(n)$ and its previously received and buffered history at delay D, i.e., $r_{STP}(n - D)$. To be more specific, the $L = 160$ samples long STP residual $r_{STP}(n)$ is divided into four $N = 40$ samples long subsegments, and for each of them one LTP is determined by computing the crosscorrelation between the presently processed subsegment and a continuously sliding $N = 40$ samples long segment of the previously received 128 samples long STP residual segment $r_{STP}(n)$. The maximum of the correlation is found at a delay D, where the currently processed subsegment is the most similar to its previous history. This is most probably true at the pitch periodicity or at a multiple of the pitch periodicity. Hence the most redundancy can be extracted from the STP residual, if this highly correlated segment is subtracted from it, multiplied by a gain factor G, which is the normalised crosscorrelation found at delay D. Once the LTP filter parameters G and D have been found, they are quantised to give G' and D', where G is quantised only by two bits, while to quantise D seven bits are sufficient. The quantised LTP parameters (G', D') are locally decoded into the pair (G'', D'') so as to produce the locally decoded STP residual $r'_{STP}(n)$ for use in the forthcoming subsegments to provide the previous history of the STP residual for the search buffer, as shown in Fig. 11. With the LTP parameters just computed the LTP residual $r_{LTP}(n)$ is calculated as the difference of the STP residual $r_{STP}(n)$ and its estimate $r''_{STP}(n)$, which has been computed by the help of the locally decoded LTP parameters (G'', D'') as shown below: $r_{LTP}(n) = r_{STP}(n) - r''_{STP}(n)$, $r''_{STP}(n) = G'' r'_{STP}(n - D)$. Here $r'_{STP}(n - D)$ represents an already known segment of the past history of $r'_{STP}(n)$, stored in the search buffer. Finally, the content of the search buffer is updated by using the locally decoded LTP residual $r'_{LTP}(n)$ and the estimated STP residual $r''_{STP}(n)$ to form $r'_{STP}(n)$, as shown below: $r'_{STP}(n) = r'_{LTP}(n) + r''_{STP}(n)$.

4 RPE computation: The LTP residual $r_{LTP}(n)$ is weighted with the fixed smoother, which is essentially a gracefully decaying band limiting low-pass filter with a cut-off frequency of 4 kHz/3 = 1.33 kHz according to a decimation by three about to be deployed. The smoothed LTP residual $r_{SLTP}(n)$ is decomposed into three excitation candidates constituted by 14, 13 and 13 samples. Then the energies E1, E2, E3 of the three decimated sequences are computed, and the candidate with the highest energy is chosen to be the best representation of the LTP residual. The excitation pulses are afterwards normalised to the highest amplitude $v_{max}(k)$ in the sequence of the 13 samples, and they are quantised by a three bit uniform quantiser, whereas the logarithm of the block maximum $v_{max}(k)$ is quantised with six bits. According to three possible initial grid positions k, two bits are needed to encode the initial offset of the grid for each subsegment. The pulse amplitudes $\beta(k, i)$, the grid positions k and the block maxima $v_{max}(k)$ are locally decoded to give the LTP residual $r'_{LTP}(n)$, where the 'missing pulses' in the sequence are filled with zeros.

The block diagram of the RPE-LTP decoder is shown in Fig. 12, which exhibits an inverse structure, constituted by the functional parts of: 1. RPE decoding, 2. LTP synthesis filtering, 3. STP synthesis filtering, 4. Postprocessing.

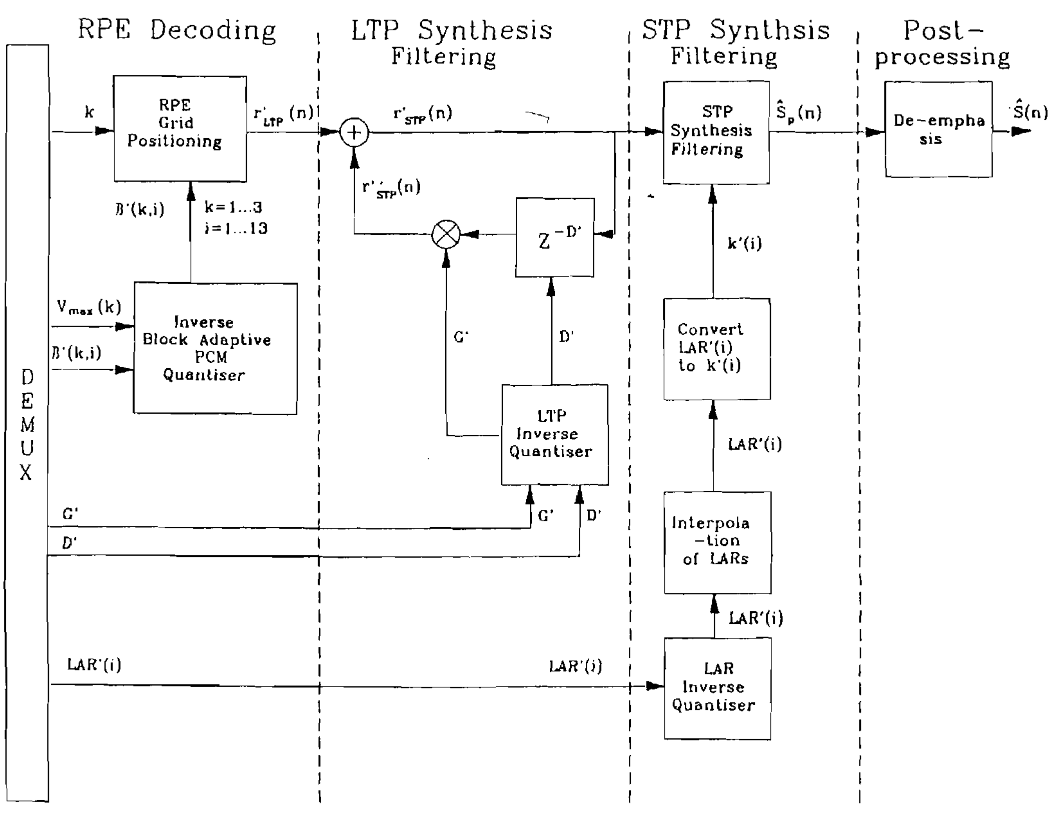


Fig. 12 - Blockdiagram of the RPE-LTP Decoder.

1 RPE decoding: In the decoder the grid position k, the subsegment excitation maxima $v_{max}(k)$ and the excitation pulse amplitudes $\beta'(k, i)$ are inverse quantised, and the actual pulse amplitudes are computed by multiplying the decoded amplitudes with their corresponding block maxima. The LTP residual model $r'_{LTP}(n)$ is recovered by properly positioning the pulse amplitudes $\beta(k, i)$ according to the initial offset k.

2 LTP synthesis filtering: Firstly the LTP filter parameters (G', D') are inverse quantised to derive the LTP synthesis filter. Then the recovered LTP excitation model $r'_{LTP}(n)$ is used to excite this LTP synthesis filter (G', D') to recover a new subsegment of length $N = 40$ of the estimated STP residual $r'_{STP}(n)$. To do so, the past history of the recovered STP residual $r'_{STP}(n)$ is used, properly delayed by D' samples and multiplied by G' to deliver the estimated STP residual $r''_{STP}(n)$, according to: $r''_{STP}(n) = G' \cdot r'_{STP}(n - D')$, and then $r''_{STP}(n)$ is used to compute the most recent subsegment of the recovered STP residual, as given by: $r'_{STP}(n) = r''_{STP}(n) + r'_{LTP}(n)$.

3 STP synthesis filtering: To compute the synthesized speech $\hat{s}(n)$ the PARCOR synthesis is used, where similarly to the STP analysis filtering the reflection coefficients $k(i)$ $i = 1 \ldots 8$ are required. The $LAR'(i)$ parameters are decoded by using the LAR inverse quantiser to give $LAR''(i)$, which are again linearly interpolated towards the analysis frame edges between parameters of the adjacent frames to prevent abrupt changes in the character of the speech spectral envelope. Finally, the interpolated parameter set is transformed back into reflection coefficients, where filter stability is guaranteed, if recovered reflection coefficients, which fell outside the unit circle are reflected back into it, by taking their reciprocal values. The inverse formula to convert $LAR(i)$ back into $k(i)$ is given by:

$$k(i) = \frac{10^{LAR(i)} - 1}{10^{LAR(i)} + 1}$$

4 Post-processing: The post-processing is constituted by the de-emphasis, using the inverse filter given below: $H(z) = 1 + c_1 z^{-1}$.

Table 3 - *Summary of the RPE-LTP bit-allocation scheme.*

Parameter to be encoded	No. of bits
8 STP LAR coefficients	36
4 LTP Gains G	$4 \times 2 = 8$
4 LTP Delays D	$4 \times 7 = 28$
4 RPE Grid-positions	$4 \times 2 = 8$
4 RPE Block maxima	$4 \times 6 = 24$
$4 \times 13 = 52$ Pulse amplitudes	$52 \times 3 = 156$
Total number of bits per 20 ms	260
Transmission bitrate	13 kbit/s

The summarised RPE-LTP bit allocation scheme is tabulated in Table 3 for a period of 20 ms, which is equivalent to the encoding of $L = 160$ samples, while the detailed bit-by-bit allocation is given in [10]. The RPE-LTP bits are classified into categories of Class 1a, Class 1b and Class 2 in descending order of prominence to facilitate a three-level error protection scheme, as explained earlier in Fig. 3.

10. CONCLUSIONS

After a short system overview this paper summarised on the basis of Recommendations [R.05.xx.] and [R.06.xx.], how speech, data and signalling messages are generated in the GSM system and highlighted the way the logical channel messages are mapped onto the physical radio-frequency channel. Channel coding and interleaving were portrayed as part of this mapping process for both information and control channels. The TDMA hierarchy of all traffic and control channels was also summarised and carrier as well as TDMA burst synchronisation were described. Part II of the paper will focus on issues of wide-band transmissions via the standardised GSM channels, considers details of discontinuous transmissions and those of some confidential algorithms as well as provides some system performance figures.

Acknowledgement

This paper is the crystallization of the GSM/ETSI standards as well as of numerous excellent papers referenced, whence the authors are indebted to all 'GSM contributors'.

Manuscript received on June 6, 1993

REFERENCES

[1] R. Steele (Ed.): *Mobile radio communication*. Pentech Press, 1992, ISBN 07273-1406-8.

[2] Proceedings of the Nordic Seminar on Digital Land Mobile radiocommunication (DMR). February 1985, Espoo, Finland.

[3] Proceedings of the Second Nordic Seminar on Digital Land Mobile Radiocommunication (DMRII). October 1986, Stockholm, Sweden.

[4] Proceedings of the International Conference on Digital Land Mobile Radiocommunication (ICDMC). June/July 1987, Venice, Italy.

[5] Proceedings of Digital Cellular Radio Conference. Oct. 12-14.1988, Hagen, FRG.

[6] A. Moloberti: *Definition of the radio subsystem for the GSM pan-european digital mobile communication system*. Proc. of ICDMC, June/July 1987, Venice, Italy, p. 37-46.

[7] A. W. D. Watson: *Comparison of the contending multiple access methods for the pan-european mobile radio systems.* "IEE Colloquium", Savoy Place, London, Digest No: 1986/95, 07.10.1986., p. 2/1-2/6.

[8] D. M. Balston: *Pan-european cellular radio: or 1991 and all that.* "Electronics and Communications Engineering Journal", jan/feb. 1989, p. 7-13.

[9] E. Natvig: *Evaluation of six medium bitrate coders for the pan-european digital mobile radio system.* "IEEE Journal on Selected Areas in Communications", Vol. 6, No 2, Febr. 1988, p. 324-334.

[10] Group Speciale Mobile (GSM) Recommendation, Apr. 1988.

[11] P. Vary, R. J. Sluyter: *MATS-D speech codec: regular-pulse excitation LPC.* Proc. of Nordic Conference on Mobile Radio Communications, 1986, p. 257-261.

[12] P. Vary, R. Hoffmann: *Sprachcodec für das europäische Funkfernsprechnetz.* "Frequenz", Vol. 42, No. 2/3, 1988, p. 85-93.

[13] J. Schur: *Über Potenzreihen, die im Innern des Einheitskreises beschränkt sind.* "Journal für die reine und angewandte Mathematik", Vol. Bd 147, 1917, p. 205-232.

[14] P. Vary, K. Hellwig, R. Hoffman, R. J. Sluyter, C. Galand, M. Rosso: *Speech codec for the european mobile radio system.* Proc. of ICASSP'88, p. 227-230.