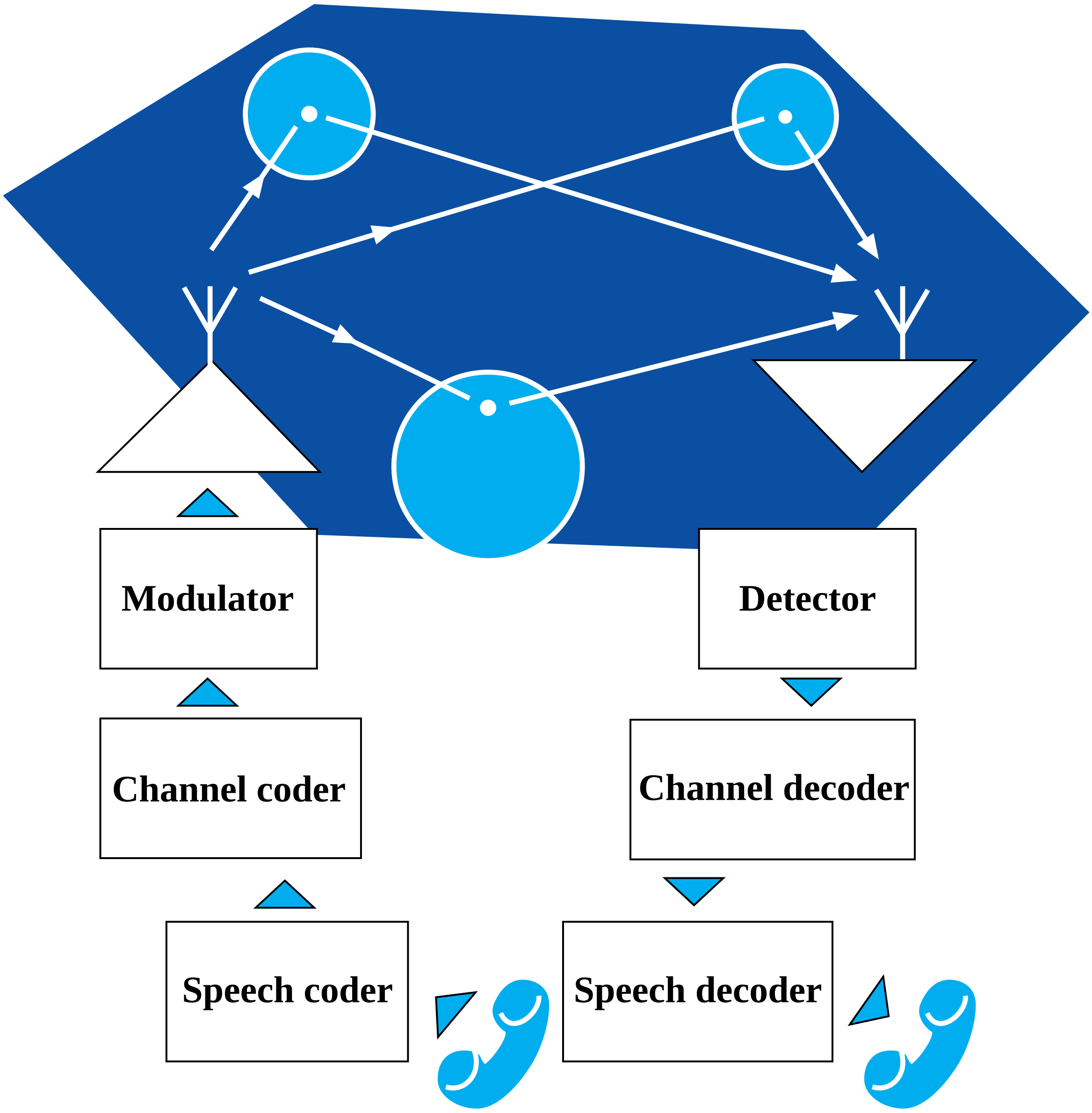
Radio School

DM2 Digital Mobile Telephony



RCUR

Core Unit Radio Systems and Technology

ERICSSON

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D-AMPS (ADC)

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1. Background

The development of digital mobile telephone systems has differed considerably in the USA as compared to western Europe. This explains the many essential differences between the Digital AMPS system (D-AMPS), previously called American Digital Cellular - ADC and the Global System for Mobile Communications (GSM).

The analog mobile telephone system, AMPS (Advanced Mobile Phone System), covers almost the entire USA and serves the needs of the majority of its users. The overriding problem is that owing to the shortage of frequencies AMPS operates since 1990 at full capacity in the major urban areas. There is short term only limited scope for increasing the traffic capacity of the system by reducing cell sizes, and there is no prospect in the near future of additional frequencies in the same band being allocated for mobile telephone services. (However similar services will be provided in a band around 1900 MHz). The principal reason for introducing a digital mobile telephone system alongside the Analog AMPS was to obtain a substantial improvement in spectrum efficiency.

Work on the specification of a digital mobile telephone system started in 1988, when the Federal Communications Commission (FCC) initiated a study phase with the stipulation that a new digital system must offer the long-term potential to provide spectrum efficiency ten times higher than that of the AMPS. The study work was coordinated by the TR-45.3 working group set up by the Telecommunications Industry Association (TIA). The work was based on a demand specification drawn up by the Cellular Telecommunications Industry Association (CTIA) and the result was the IS-54 standard – the Digital-AMPS.

Two system solutions were presented to the working group: narrowband FDMA (N-FDMA) and narrowband TDMA (N-TDMA). The starting point for both was that a threefold improvement in spectrum efficiency should be obtained in the first phase by accommodating three speech channels in a 30-kHz radio channel. Two variants of N-FDMA were submitted: one by ATT/Bell and the other by Motorola. The N-TDMA system was submitted by Ericsson. Pilot systems were developed and demonstrated in field tests. From the results of these, the working group drew up a recommendation for N-TDMA, followed by a detailed standard specification: EIA/TIA-IS-54.

The D-AMPS system is complementary to the analogue AMPS, the two systems forming an integrated system. The digital system is gradually being introduced in the main urban areas. System capacity is being increased by replacing an analog 30-kHz base-station unit for one speech channel by a digital unit for a 30-kHz radio channel with TDMA for three speech channels. Each time a unit is exchanged in a cell, the total number of speech channels in the system is increased by two.

The analog system alone will continue to serve the rural areas for a long time yet, which means that mobile terminals will have to switch between the the analog and digital system to make full use of the integrated system both inside and outside the main urban areas. The D-AMPS system also depends on the analog system for most of the signalling, e.g. for setting-up calls and allocating traffic channels. The signalling complexity of the D-AMPS system is therefore much lower than in the GSM.

Initially, the D-AMPS network will constitute isolated islands within an otherwise analog system. It must be possible to exchange an analog base station in an individual cell for a digital one, without the need to change the overall frequency planning. This would hardly be possible if the system were based on FDMA, whereby a 30-kHz analog channel would be replaced by three digital 10-kHz channels. Because these would end up in the same cell, adjacent channel selectivity approaching 70 dB would be required, which is incompatible with existing analog systems having cell structures based on an interleaved channel plan. This is because this arrangement permits adjacent channel selectivity of only 30-40 dB, since adjacent channels are not allocated to the same cell or adjacent cells. The problem presented by the high requirement for adjacent channel selectivity using 10-kHz channels is avoided with the use of TDMA. This is one of the principal reasons that N-TDMA was chosen instead of N-FDMA.

The TDMA arrangement accommodates three speech channels in a 30-kHz channel. Continued advances in speech coding could eventually result in a further doubling of the capacity (six speech channels in a 30-kHz frequency slot). As in the GSM, the system solution for the D-AMPS system therefore enables each carrier to carry twice the number of half-rate traffic channels. Another way in which the spectrum efficiency can be enhanced in the long term is through tighter geographical packing (smaller cell clusters). Thus, the long-term potential of the D-AMPS system is to improve spectrum efficiency more than tenfold as compared to that of the Analog AMPS. To start with, however, the digital system will have to be adapted to the same cell structure as in the analog system.

Development work is continuing in two areas. First, the digital system is to be augmented by a complete arrangement of signalling channels plus control and monitoring equipment, which will enable it to be installed as an autonomous radio system, without support from the Analog AMPS, see appendix 2. Second, it is planned to use a higher frequency band at around 1900 MHz. This will free the system from the cellular planning of the analogue mobile telephone system, allowing smaller cluster sizes, provided that the D-AMPS system can be augmented to cope with a lower protection ratio than that required by the analog AMPS.

Another consequence of the frequency planning being closely aligned with that of the analog system is that, unlike in the GSM, frequency hopping cannot be used. The narrow modulation width also implies that no appreciable gain can be obtained from multi-path diversity based on suitable channel equalization. It is possible that channel equalization will only be necessary to overcome particularly difficult propagation conditions (such as in mountainous terrain). The scope for introducing interleaving is also limited because of the relatively long duration of a TDMA frame. This means that if reasonable diversity gain against fading due to multipath propagation is to be achieved in quasistationary connections, antenna diversity will be essential. (However, this does not give any improvement in spectrum efficiency so long as the cluster size is determined by the analog system.)

2. Radio specification

To accommodate three speech channels on a 30-kHz radio channel imposes heavy demands on the subsystems involved – speech coders, channel coders, radio modem and TDMA formatting. Advances made since the GSM specification was finalized enabled the data rate from the speech coder to be reduced from 13 kb/s in the GSM to around 8 kb/s in the D-AMPS system. The coding rate is also higher in the D-AMPS system than in the GSM.

The bandwidth expansion on modulation is considerably less than in the GSM, thanks to the use of linear Nyquist-filtered modulation. The drawback is that the transmitter output stage must have good linearity characteristics.

To reduce the relative overhead in TDMA formatting, longer time slots are used than in GSM. This means a smaller increase in the system data rate to accommodate guard slots and synchronization sequences. The drawback with long time slots is not only that channel equalization must follow the changes in the impulse response of the propagation channel during a data burst (rendering channel equalization more complex) but also that the possible interleaving depth will be insignificant (otherwise the transmission delay would be too long).

A summary of the principal radio parameters is given in Fig. 2.1. Since linear modulation is used and the requirement for adjacent channel selectivity is only moderate, a 48-kb/s system data rate can be used over a 30-kHz channel. The $\pi/4$ arrangement reduces variations in the signal envelope which, in turn, reduces the requirement for linearity in the transmitter amplifier. Nonetheless, a fairly complex transmitter amplifier with special arrangements to improve linearity is required.

Transmission specification for the USA Digital MTS (D-AMPS/IS-54)

Frequency band: (frequency duplex)	825–850 MHz 870–895 MHz
Channel spacing	30 kHz
Modulation	π / 4-DQPSK (4QAM)
System data rate	48.6 kb/s
TDMA	Frame 40 ms
Time slots	6 x 6.67 ms
Full-rate channel data rate	13 kb/s
Speech coder	Code-excited LPC (CELP) also called Vector-sum excited LPC (VSELP) 7.95 kb/s
Diversity	(Antenna) Channel coding Interleaving

Fig. 2.1

The **TDMA structure** is shown in Fig. 2.2. As in the GSM, the timing of the inward and outward structures differs to make time duplex possible (the received and transmitted time slots in the terminal occur at different times).

In addition, the base instructs the terminals of suitable delays between incoming and outgoing data bursts, in order to offset different delays in propagation times between terminals close to and remote from the base station. The procedure is basically the same as in the GSM.

In some situations, a terminal will need to transmit before it has received an instruction from the base station as to the delay setting it should use. To eliminate the risk of the transmitted data burst covering two time slots at the base-station receiver, a reduced length data burst is used. This is also similar to the procedure in the GSM.

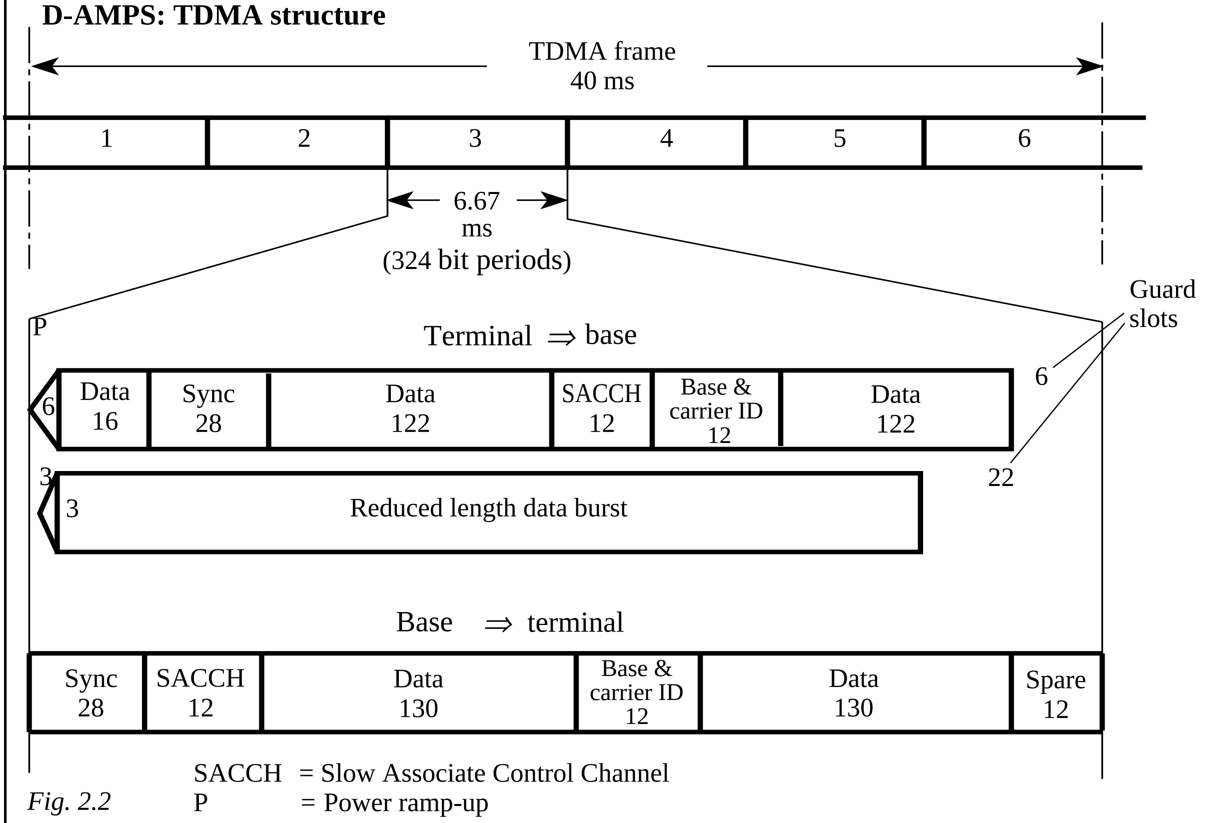


Fig. 2.2

To facilitate a subsequent transition to half-rate channels, a TDMA frame is divided into six time slots. Initially, two of these are used for each full-rate channel, which means that three traffic channels are available. The time slots are handled somewhat differently in the two directions. In the inward direction, as in the GSM, guard slots are interposed between data bursts in adjacent time slots, and short intervals are used for ramp-up and ramp-down of the transmission pulse. These intervals are not required in the outward direction (TDM instead of TDMA), since the transmitter will not be pulsed and the data blocks will be packed close together without any intervening guard slots.

Each time slot carries an information sequence, **system signalling** (Slow Associate Control Channel, SACCH) and an ID code for the base station/operator and carrier. A synchronization sequence is also needed, which is also used for setting the channel equalizer. Each time slot contains 260 traffic bits; since 25 frames per second are transmitted and two time slots per frame are used for a full-rate channel, the gross data rate for a traffic channel will be 13 kb/s (260 x 25 x 2). A Fast Associate Control Channel (FACCH), which is needed for fast signalling during handover, is obtained by ”stealing” time frames from the speech signal (compare the GSM).

The **channel equalization** requirements differ greatly between the D-AMPS system and the GSM. A larger equalizer window is specified for the D-AMPS system (40 µs), which corresponds to the length of only one symbol period. On the other hand, the channel equalizer has to be adapted to variations in the impulse response of the radio channel during a data burst. If a Viterbi-type channel equalizer is used, the number of states will be four, since the modulation type employed is 4-QAM. For each state (node), there are four entry and exit paths through the trellis. The level of complexity is roughly the same as in the GSM.

Linear Nyquist-filtered modulation is used to minimize the modulation bandwidth, which complicates the transmitter output amplifier. To reduce the linearity requirement for the amplifier, a phase shift of $\pi/4$ is introduced between each symbol (see Fig. 2.3).

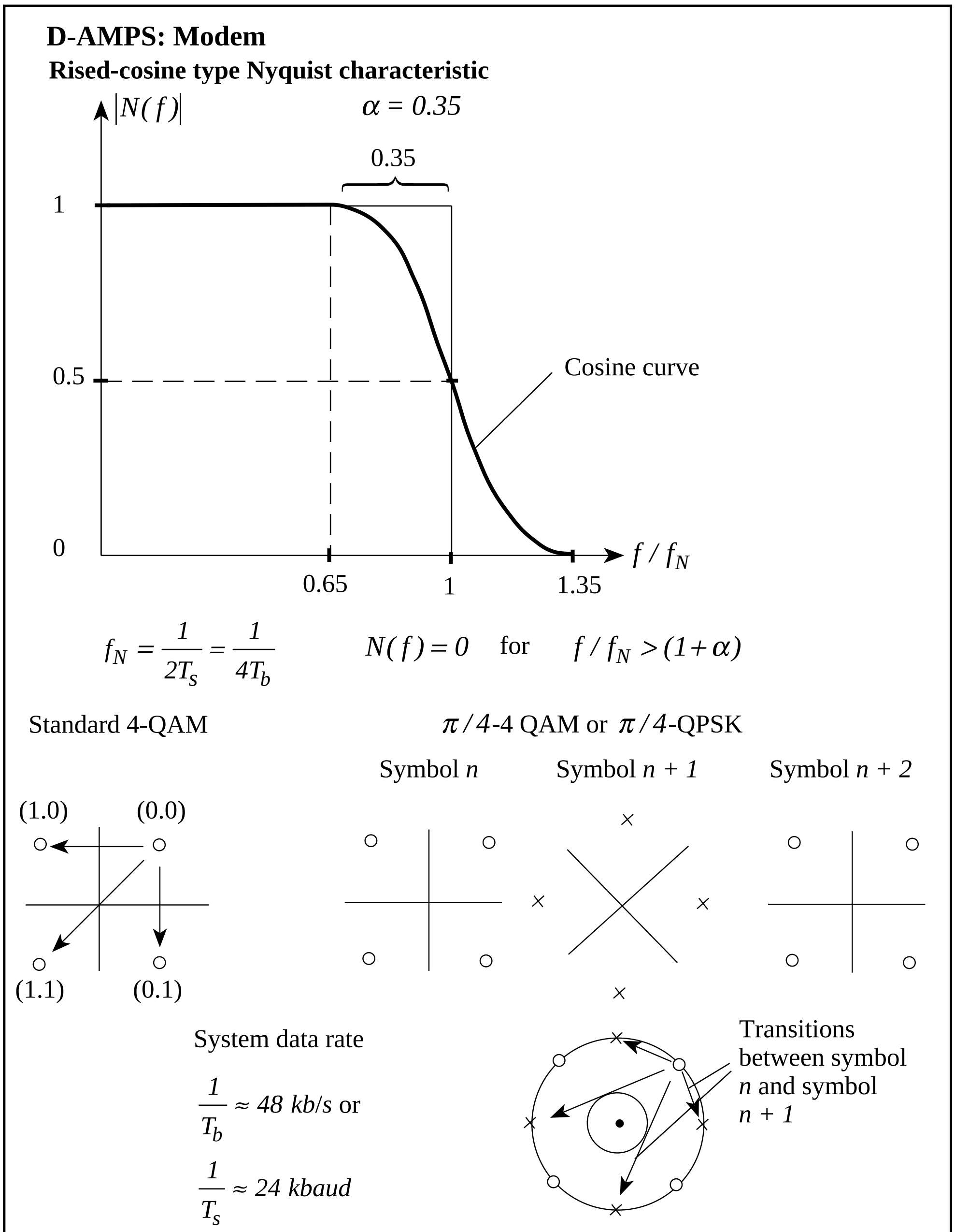


Fig. 2.3

Thus, the chance of the signal envelope reaching the zero level in some symbol transitions is averted. This implies a less stringent requirement for linearity on the transmitter amplifier, since class B or B/C output stages have strong nonlinearity at low power level (see Fig. 2.4). For the Nyquist characteristic, *N(f)*, a Rised Cosine has been selected, with the value of the α parameter being 0.35. To ensure matched receiver conditions, the same selectivity of the transmitter filter and the receiver filter has been introduced: $|N(f)| = |T(f)R(f)|$, that is $|T(f)| = \sqrt{|N(f)|}$.

The modulation spectrum is thus of the Root Rised Cosine type, with $\alpha = 0.35$. In the case of 4-QAM with a bit data rate ≈ 48 kb/s, the Nyquist frequency, f_n, ≈ *12 kHz*, which corresponds to a 3-dB bandwidth of $2f_n \approx 24$ *kHz*. In theory, there is infinite attenuation of the modulation spectrum outside a bandwidth of $2(1+\alpha)f_n \approx 32$ kHz. The spectrum therefore has steep flanks with little power outside the 30-kHz channel width.

Amplifier stage. Modulation curves

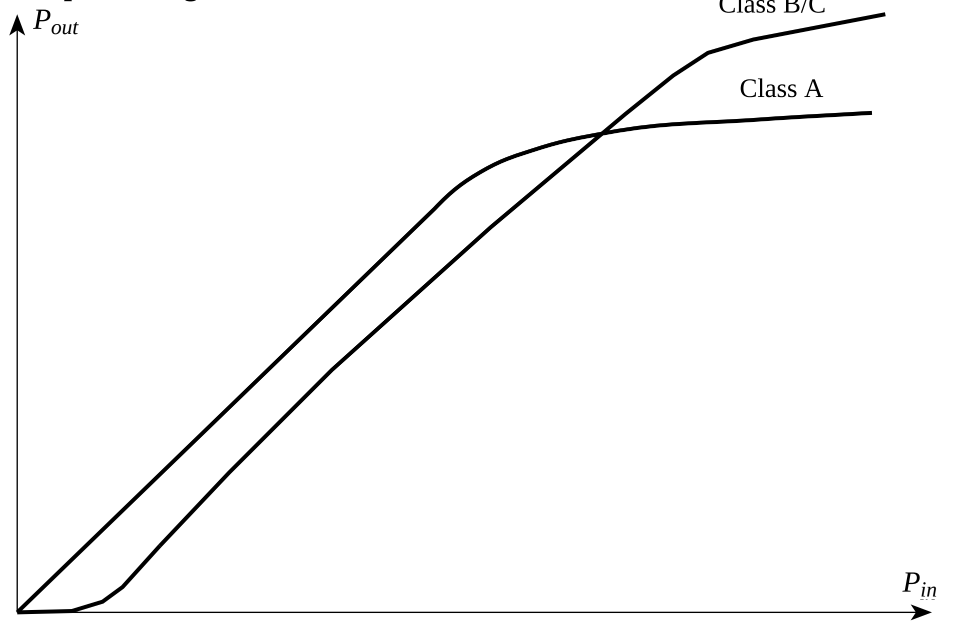
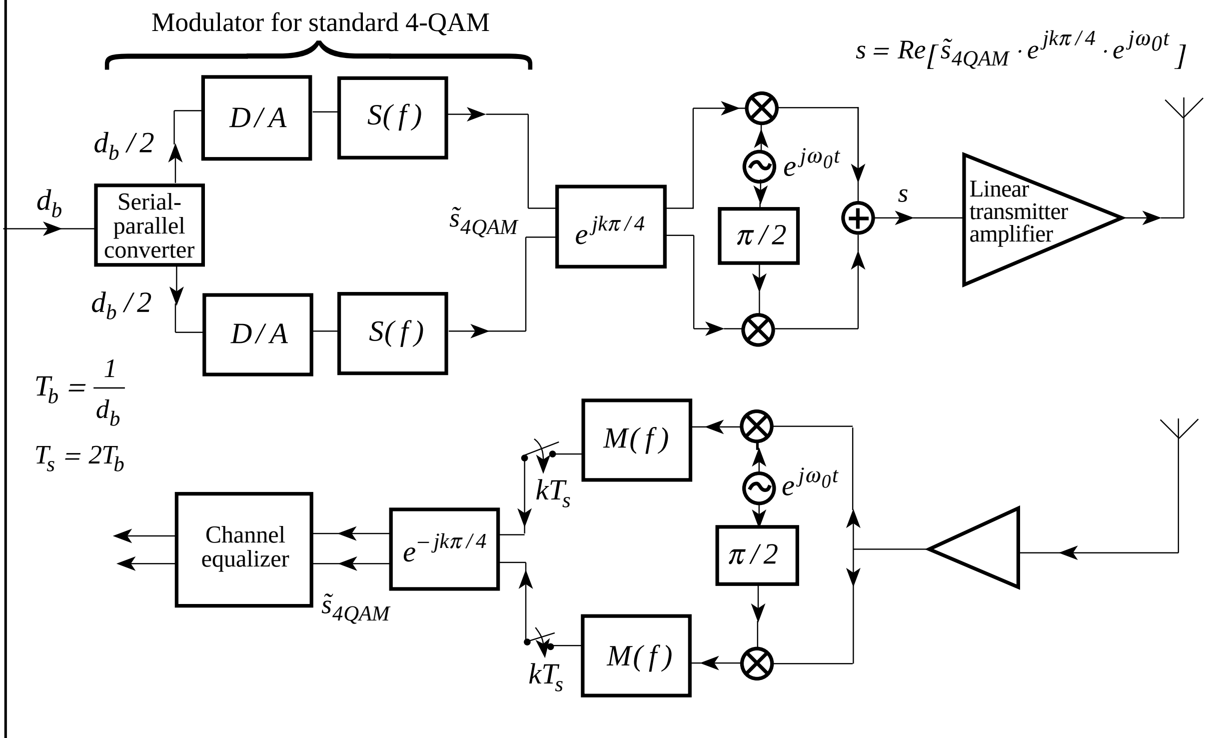


Fig. 2.4

D-AMPS: π / 4-DQPSK modulation



Nyquist characteristic: $|N(f)| = |S(f)| \cdot |M(f)|$ (Compare Fig. 3.10 in Module DM1)
(Differential coding omitted)

Fig. 2.5

The phase shift of *π/4* between each symbol is introduced in the output from the 4-QAM modulator. A compensating phase shift of -*π/4* per symbol is introduced before the channel equalizer, so that the input signal to the channel equalizer is a standard 4-QAM signal (see Fig. 2.5).

Differential modulation has been specified so that the receiver in principle can be realized without the need for a phase-locked local oscillator. This means that the 4-QAM modulator is equipped with a precoder in the *I* and *Q* channels, the output signals from which correspond to the difference between the current bit and the preceding bit. For detection, a decoder has been introduced that gives the inverse function to that of the precoders. However, because the D-4QAM signal format is unsuitable for the channel equalizer, transition to D-4QAM takes place after it.

The difference in the data rate from the speech coder (7.95 kb/s) and 13 kb/s is exploited for **channel coding**. The principle is the same as in the GSM (see Fig, 2.6). In the D-AMPS, the bits are also grouped into three classes according to their sensitivity to transmission errors.

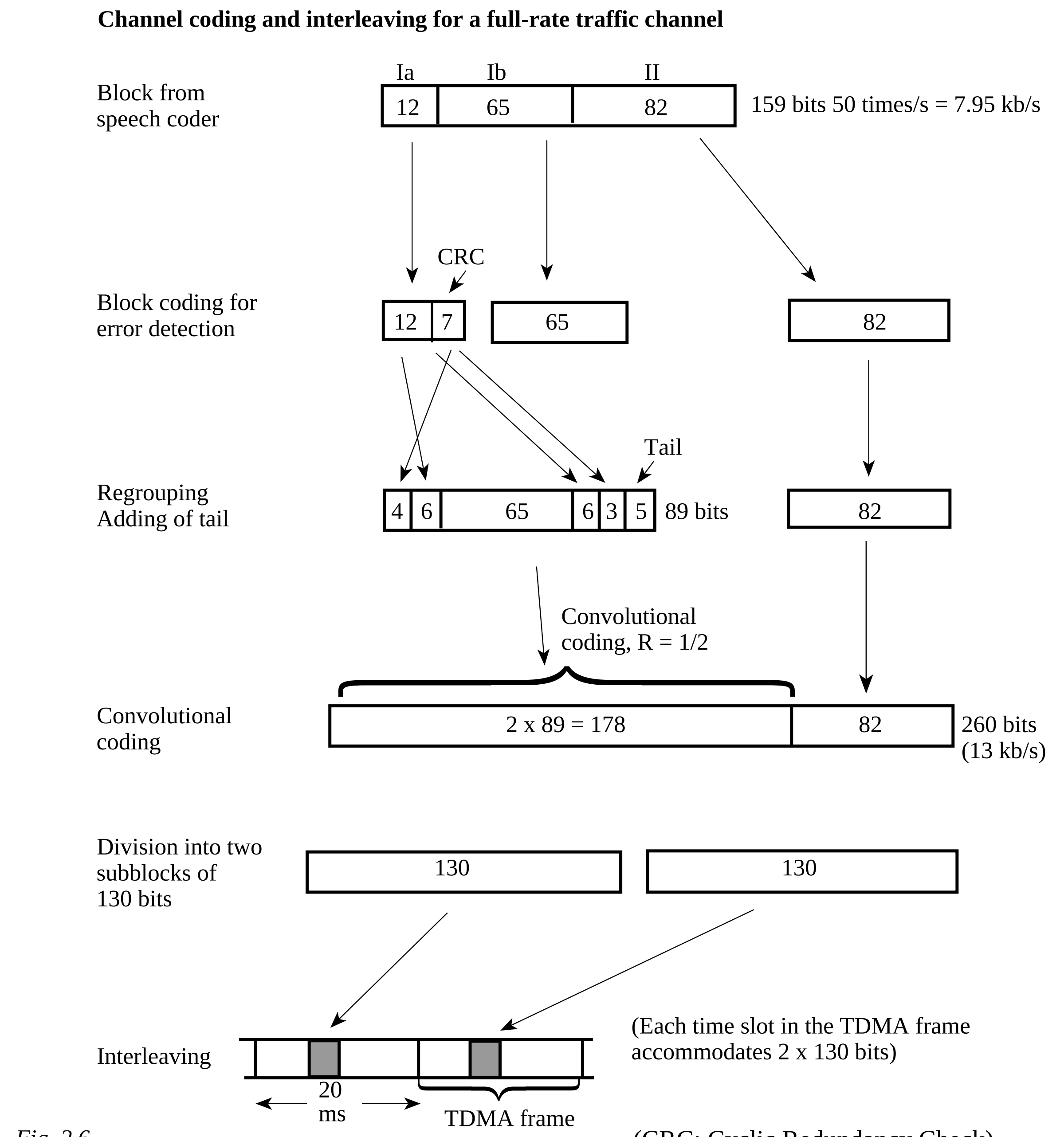


Fig. 2.6

(CRC: Cyclic Redundancy Check)

The output signal from the speech coder occurs in data bursts of 159 bits every 20 ms. (Speech coding is discussed elsewhere.) When the signal is passed through the channel coder, the number of bits is increased to 260. The channel coder is adapted to the characteristics of the speech coder such that only certain sensitive bits (77) are protected by FEC channel coding (convolutional code with R = 1/2 and constraint length 5). Twelve of these 77 bits are very critical and are therefore also protected by error detection coding (seven parity bits). If this group of 12 bits should be subject to transmission errors, special steps will be taken on the receiver side, such as repeating the preceding speech frame. 5 tail bits are added so that the shift register in the encoder is reset to zero at the end of each burst. After channel coding, interleaving takes place over two time slots 20-ms apart.

The **diversity gain** from channel coding is much lower than in the GSM, at a low terminal speed, as the D-AMPS system has few support facilities for the channel coding. Interleaving is insignificant and no frequency hopping is utilized to break up stationary fading structures.

In the D-AMPS system, **antenna diversity** therefore would give a considerable reduction in the required fading margin for multipath propagation. The proposed antenna diversity arrangement at the terminals is based on the possibility offered by TDMA to listen to the received signal from both antennas during free time slots occurring before the receiving time slot. Thus the best antenna can be connected well before the receive time slot. Therefore the switching transients disappear before the start of the receive time slot.

It has been estimated that the required protection ratio is 16 dB without, and 11 to 12 dB with, antenna diversity. Even without it, the protection ratio for the D-AMPS system is slightly better than that of the existing analogue system. There is therefore no obvious gain from employing antenna diversity so long as the D-AMPS system is forced to use a cell structure that is entirely determined by that used in the analogue system.

3. Comparison of the GSM and D-AMPS system

The GSM and the D-AMPS system need to satisfy roughly the same communication needs. They are both cellular mobile telephone systems having extensive coverage, serving both rural areas and major population centres. Because of the need to coexist with the established analogue mobile telephone system in the same or adjacent bands, both systems must employ frequency duplex. Both the GSM and the D-AMPS system also have to meet the same requirements in terms of transmission delay in speech transmission. Speech is the dominating application, although the need to interface with ISDN was more of a consideration at the design of the GSM.

However, the background conditions for the systems are otherwise quite different, as a result of which there are essential differences in numerous system parameters. The main differences are given in Fig. 3.1.

Comparison of the GSM and D-AMPS system, A

GSM

New frequency band allows wide freedom in selection of channel spacing

Total spectrum efficiency essential (combination of bandwidth requirement and geographical packing density)

D-AMPS

Channel spacing same as in the analogue system

Need to minimize bandwidth requirement per speech channel (geographical packing density determined by the existing analogue system)

Basic specification finalized two years later then the GSM's (more advanced technological solutions)

Fig. 3.1

The different requirements and design criteria have also resulted in essential differences between the systems in the TDMA structure and signal processing (see Figs. 3.2-3.4).

Comparison of the GSM and D-AMPS system, B

a. GSM

Maximum system data rate and length of time slots determined by need for reasonable complexity of channel equalizers.

- No adaptation during a TDMA time slot, i.e. short data bursts required (0.58 ms)
- Equalizer able to cope with time dispersion having a maximum length of four symbol periods (16 μs)

Comprehensive signal processing justified to reduce cluster size (channel coding, 4.6 ms frame time giving interleaving depth of 8, frequency hopping)

b. D-AMPS

Maximum system data rate determined by channel spacing (linear modulation and 30-kHz channels give approx. 50 kb/s)

Necessary with small reduction in capacity due to guard slots and synchronization sequence. Therefore need for long time slots (6.7 ms)

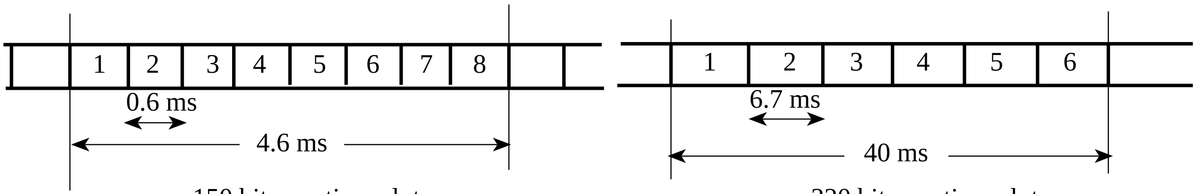
Cluster size determined by existing cell structure. Acceptable with protection ratio only moderately better than in the analogue system (reduced channel coding, small interleaving depth, no frequency hopping)

In future, reduction in cluster size is possible through antenna diversity also in the terminals

Fig. 3.2

Comparison of the GSM and D-AMPS system, C

TDMA structure: GSM | **TDMA structure: D-AMPS**



≈ 150 bits per time-slot

Frame time determined by terminals' needing to be tuned to three frequencies: transmit, receive and listening (≈ 5 ms).

The time-slot shall be short enough (≈ 0.5 ms) to permit fixed equalizer setting .

≈ 320 bits per time-slot

Long time-slot period owing to:
- long time per radio symbol
- low relative TDMA overhead requires many bits per time-slot

2 x 3 = 6 time-slots to allow subsequent transition to half-rate speech coders.

Time-slot are so long that channel equalizers must be adapted during each time-slot, although equalizer window of only one symbol period is needed.

Fig. 3.3

Comparison of the GSM and D-AMPS system, D

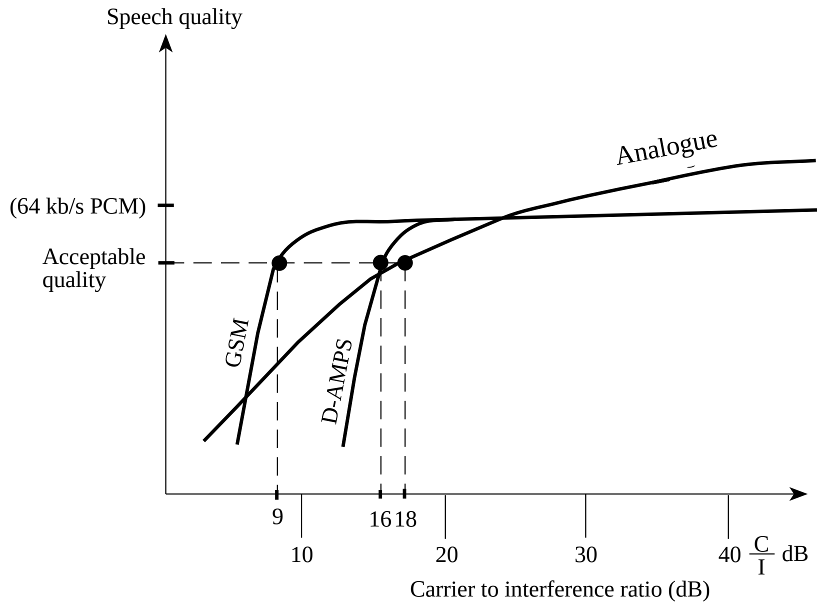


Fig. 3.4

A summary of the most important radio transmission parameters for the GSM and D-AMPS system is given in Fig. 3.5.

Comparison of the GSM and D-AMPS system, E

	GSM	D-AMPS
Channel spacing	200 kHz	30 kHz
Channel width/half-duplex speech channel	25 kHz	10 kHz
Modulation	GMSK	QAM (π/4-DQPSK)
System data rate	271 kb/s	48.6 kb/s
TDMA frame	4.6 ms	40 ms
Time slots	8 x 0.57 ms	6 x 6.67 ms
Bit rate, full-rate speech channel, net (with channel coding), gross	13 kb/s 22 kb/s	7.95 kb/s 13 kb/s
Rate (channel coding)	0.57	0.68
Bandwidth expansion*	200/(8 x 13) = 1.92	30/(3 x 8) = 1.25
Interleaving depth (full-rate speech channel)	8	2
Frequency hopping	Possible	No

** Channel spacing per speech channel divided by the data rate from the speech coder*

Fig. 3.5

Comparison of the GSM and D-AMPS system, F

	Analogue FM (NMT 450)	GSM	D-AMPS without antenna diversity	D-AMPS with antenna diversity
Number of speech channels/25 MHz	1000	1000	2500	2500
Protection ratio	18 dB	9 dB	16 dB	11dB
Cluster size	3 x 7	3 x 3	3 x7	3 x 4
Speech channels/cell	47	111	119	208
Capacity improvement factor	1	2,4	2,5	4,4

Fig. 3.6

A summary of the principal radio network parameters is given in Fig. 3.6. The D-AMPS system requires a much smaller frequency slot per speech channel. On the other hand, the basic version without antenna diversity requires a considerably higher protection ratio than the GSM. This in itself is not significant, so long as the cluster size is determined by the requirement for integration with the AMPS. The total spectrum efficiency (speech channels per cell per MHz) is much the same as for the GSM. However, a stand-alone D-AMPS system with antenna diversity (cluster size of 3 x 4) has significantly better spectrum efficiency than the GSM. (For the corresponding Japanese system has been specified antenna diversity also at the terminals).

A general comparison of the background conditions for the GSM and the D-AMPS system is shown in Fig. 3.7.

Comparison of the GSM and D-AMPS system, summary

GSM

- New frequency band allocated (although a part of the band is for a limited time used by analogue MTS)
- No strict requirements on narrow bandwidth per speech or traffic channel
- Width of radio channel not critical (although inappropriate with very wide radio channels)
- Technical risk too high for introduction of modulation with varying signal envelope

The GSM achieves excellent geographical spectrum efficiency (small cluster size)

D-AMPS

- The D-AMPS system is introduced gradually in a frequency band that is already in use and within the existing cell structure for analogue MTS (AMPS). Channel width therefore predetermined at 30 kHz (or submultiple).
- The analogue AMPS is to be retained for a long time in rural areas. Principal reason for the D-AMPS: improved spectrum efficiency in urban areas
- TDMA not really justified with fewer than three speech channels per radio channel (carrier)
- FDMA cannot meet requirement for adjacent channel selectivity
- Principal problem: How to accommodate TDMA with three time slots in a 30-kHz radio channel.

The D-AMPS system achieves excellent spectrum efficiency in the frequency domain but poor geographical spectrum efficiency (unless antenna diversity introduced).

Fig. 3.7

Appendix 1: Personal Digital Cellular system (PDC)

The Personal Digital Cellular system (PDC) system (previously called Japan Digital Cellular or Pacific Digital Network) was specified by the Japanese telecommunications authority (NTT) after extensive technological development work and studies. It is similar to the D-AMPS system but in some respects more advanced and, consequently, provides higher spectrum efficiency.

The PDC system has much the same TDMA structure as D-AMPS, with a TDMA arrangement allowing three speech channels per carrier. As in the GSM, free time between transmit and receive time slots is used by the terminals to listen to carriers from adjacent cells (Mobile Assisted Handover). In principle, the same arrangements for FACCH and SACCH are used as in the D-AMPS system. The system also includes provision for changing to half-rate traffic channels later on.

An important difference between the PDC and the D-AMPS systems is that in the PDC system advanced antenna diversity (Post Detection Diversity) at base and terminals constitutes an integral part of the system solution. This has influenced the design of the system in two respects:

a) Antenna diversity gives moderate suppression of the time dispersion over the radio channel. This together with a suitable antenna arrangement on the base-station side has proved to be adequate to cope with the effect of time dispersion without the need to introduce channel equalization. It is also likely that the propagation conditions with respect to time dispersion are less extreme in Japan than in the USA.

b) Antenna diversity reduces the need for channel coding to bridge the fading dips caused by multipath propagation. Channel coding in the PDC uses a lower rate than the D-AMPS. This compensates for the need to employ a lower system data rate, owing to the narrower channel spacing (25 kHz in Japan as against 30 kHz in the USA). Otherwise, the speech coders, the speech coder data rate and the modulation type are largely equivalent to those in the D-AMPS. The post channel coder data rate in the PDC is 11.2 kb/s as against 13 kb/s in the D-AMPS.

The spectrum efficiency of the PDC is better than in the D-AMPS (without antenna diversity) partly because of the narrower channel spacing and partly because of the lower protection ratio (≈ 13 dB). This allows a cluster size of 3 x 4. NTT has also developed an advanced base-station antenna, which allows the tilt of the antenna lobe below the horizontal to be adjusted individually for each base-station site. This reduces the average co-channel interference and can also reduce strong reflections from remote objects, i.e. reduce the delay spread.

Another development by NTT is highly linear transmitter amplifiers with good efficiency based on adaptive feed-forward, which enables simultaneous amplification of a large number of carriers with 70-dB intermodulation suppression. These amplifiers have also been introduced in the existing analogue system. The result is a reduction in size of the analogue base-station equipment sufficient to provide the space required by the new digital system within existing buildings.

The type of modulation employed ($\pi/4$-DQPSK) also requires linear transmitter amplifiers in the terminals. For these to achieve adequate efficiency (40%), a dynamically variable supply voltage for the amplifier output stage is used (Linearized Saturation Amplifier with Bidirectional Control). This means that the supply voltage is adapted to variations in the level of the signal envelope. High transmitter efficiency reduces the need to conduct away dissipated power and thus facilitates extreme miniaturization.

Reference

R.W. Lorenz: Vergleich der digitalen Mobilfunksysteme in Europa (GSM) und in Japan (JDC) unter besonderer Berücksichtigung der Wirtschaftlichkeitaspekte.

Appendix 2: D-AMPS with Digital Control Channel

1. Introduction

The original specification for D-AMPS (IS-54B) was based on the concept of system integration with the previous analog AMPS. Analog AMPS was used for all the signalling before a call was set up on a digital traffic channel within the D-AMPS part of the system. To make a fully digital system possible which is completely independent of the analog AMPS, a new specification IS-54C has been established. This appendix gives a short summary of the most important characteristics. A more detailed description can be found in Ericsson Review No 2, 1994: A New Standard for North American Digital Cellular.

The main new feature of IS-54C is the introduction of a digital control channel, DCC. On the same time additional requests from the operators (User Performance Requirements for a Digital control Channel issued by Cellular Telecommunication Industry Association) have been satisfied. The main new operational features are:

- Support for microcell operation, incl. incorporation of private networks (R-PABX)
- Sleep mode provision for idle mobiles to enhance battery life time
- Increased control channel capacity and flexibility, incl. support of new data services, i.e. asyncronuous data, group 3 fax and especially short message service, SMS.

The DCC is based on the same transmission structure as used by the traffic channels within D-AMPS. The same modulation, system data rate and basic TDMA frame are used. One of the radio channels in each cell accomodates the DCC, which replaces one of the digital traffic channels (DTC). See Fig. 1a.

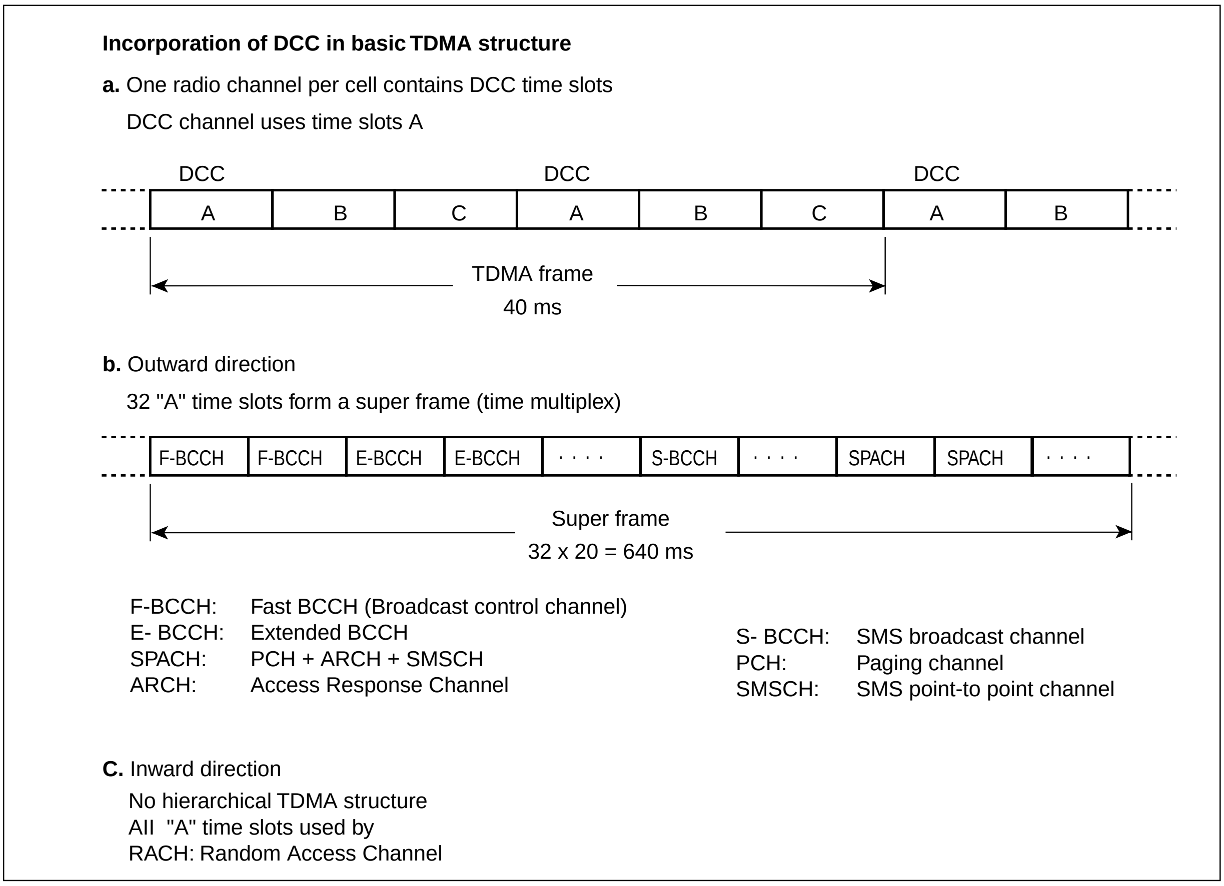


Figure 1

As in GSM, the DCC shall perform the following basic functions:

- Guide the mobiles to lock on to the radio channel comprising the DCC
- Syncronize to the DCC, incl. frame syncronization
- Broadcast messages about the network structure
- Registration of mobiles
- Location updating
- Paging mobiles to initiate the setting up of a traffic channel
- Handling requests from mobiles of channel allocation for system signalling or for traffic.

2. Description of the DCC

2.1 Burst structure

The structure of a DCC burst in a TDMA time slot is fairly similar to a traffic burst, see Fig. 2. Additional data fields in the outward direction are **SCF** and **CSFP**. The SCF is the return channel in ARQ arrangements which improves the performance of the inward paging channel. The CSFP field makes it possible for the mobiles to syncronize to the base station timing of the TDMA hierarchy (basic TDMA frame and superframe). In the inward direction a **PREAM** field is included. This field contains no information. Its purpose is to give the base receiver time to adjust the AGC to the level of the incoming burst.

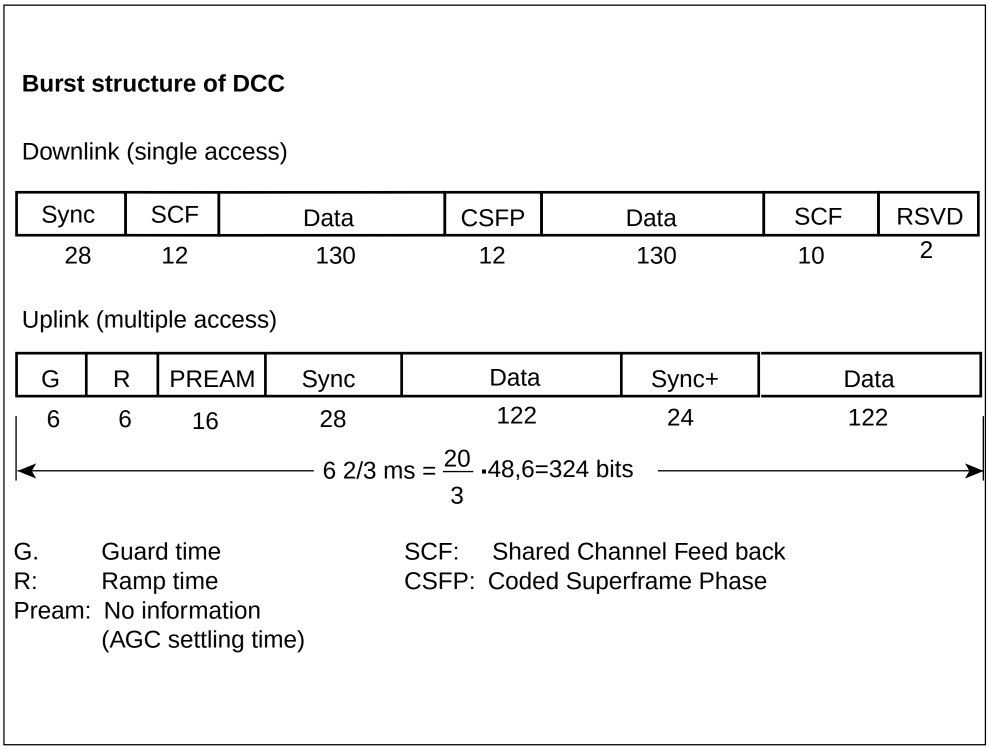


Figure 2

2.2 TDMA/TDM structure

In the inward direction, all the DCC slots are used as a universal paging channel (PCH), see Fig.1c. It can operate either as a contention channel (slotted Aloha) or on reservation basis. Immediate acknowledgement is given by the base on the SCF channel, so that immediate repetition of of the page can be made, if the first page was not received successfully.

In the outward direction, many different types of logical channels must be set up. Therefore the DCC slots in 32 basic TDMA frames form a superframe, which can be considered as a TDM arrangement with 32 channels, see Fig. 1b. In the first two slots in each superframe is placed the F-BCCH channel, which contains the most essential broadcast information, which must be repeated often. The channel also informs the mobiles how the other slots in the TDM arrangement are utilized. Additional broadcast data is transmitted over the EBCCH, which is organized in a way that permits several repetition rates. The number of slots allocated for E-BCCH can vary. Next follows a number of slots for the S-BCCH, which contains SMS messages of the broadcast type. The remaining slots in the TDM arrangements are used for a combined channel "SPACH", which is used for the outward paging channel (PCH), the access response channel, ARCH for point-to-point outward signalling and the SMSCH channel (point-to point SMS).

2.3 Hyperframe, paging classes, SMS frames

In order to increase the chance of success of an outward page over the fading radio channel, each page is sent in two successive superframes, which form a hyper frame, see Fig. 3. After two paging attempts, there might be a delay of several hyper frames before there is a new possibility (hyperframe) to page a certain mobile. There are 8 paging classes corresponding to different time intervals between the hyper frames assigned for paging of a certain group of mobile. Class 1 gives the possibility to send a page every hyperframe, class 2 allows pages to a certain group of mobiles every second hyperframe and class 8 allows for paging every 96 hyperframe. The mobiles knows in which hyperframes they might receive pages, so that they can go to sleep during the other hyperframes. If too many page requests arrive at a paging slot, the overflow is handled by a later paging slot.

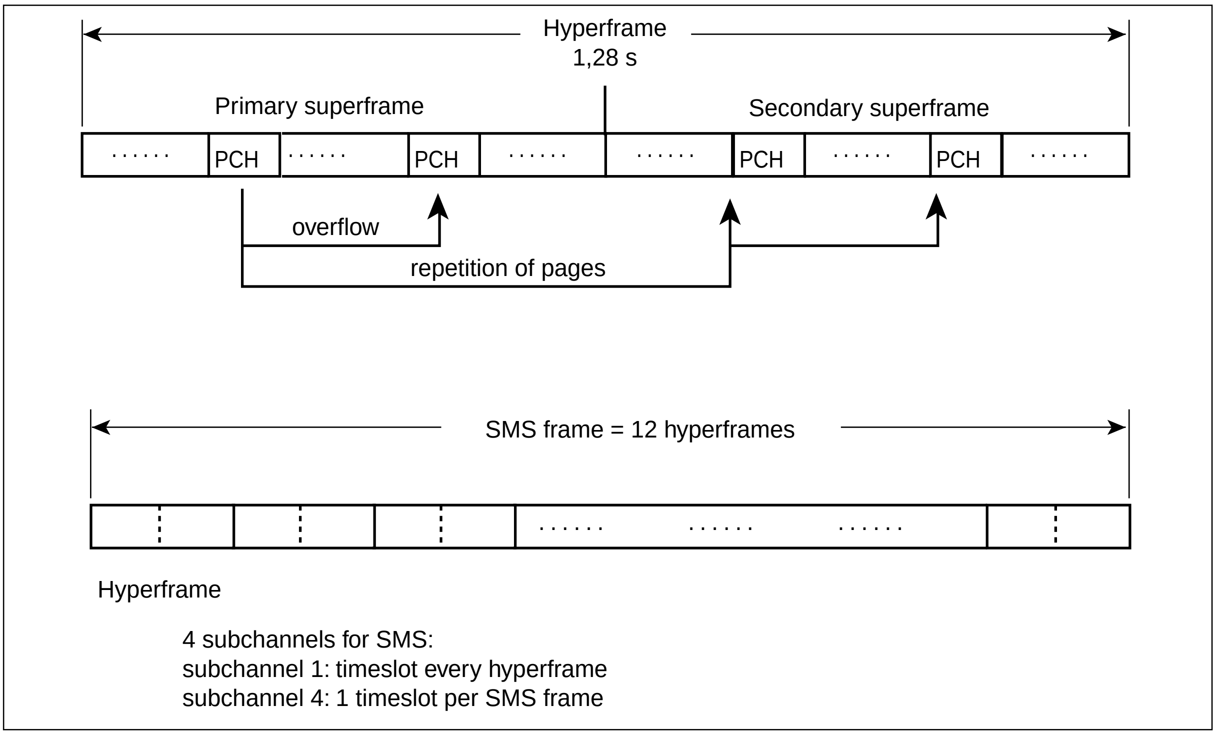


Figure 3

The SMS-frame applies to the S-BCCH (used for sending broadcast SMS messages). Different repetition rates are suitable for different types of SMS messages. The repetition time can be chosen by allocating the messages to 4 subchannels with different repetition rates. A SMS frame consists of 24 superframes.

3. New system features

3.1 SMS (Special Message Service) to mobiles

SMS of the broadcast type has been discussed above. SMS messages of the point-to-point type are sent either over the traffic channels or over the DCC on the SMSCH. The messages can be up to 239 characters long. The longer messages require several time slots, and an ARQ arrangement might be used.

3.2 Hierarchical cell structures, MACA

Mobile-assisted channel allocation (MACA)

When a connection is set up on a digital traffic channel, the MAHO is based on measurements by the mobile terminal of the signal levels from surrounding cells. This has already been implemented in the original D-AM PS according to IS-54B. Similar arrangements are implemented for the DCC, i.e. when the mobiles are in the idle mode. The base station sends a neigbor list on the BCCH, which informs the mobiles where to look for potential cell reselection. At system access (call origination) the mobile sends over the measured signal levels on the channels indicated in the neigbor list.

Hierarchical cell structure

The hierarchical cell structure is based on dedicated frequency bands for different cell types, e.g. macro cells, public micro cells and private micro cells. Due to frequency economy considerations, it is generally desirable to allocate traffic to the smallest cell, whenever there is a choice. IS-54C provides two mechanisms for forcing down the traffic to the micro cell layer. See Fig. 4. One mode selects a preferred microcell, whenever the signal level from the micro cell exceeds a specified minimum level. The other mode selects a micro cell if the difference between the measured signal levels for the macrocell and the microcell is less than a specified off-set value (biased cell selection).

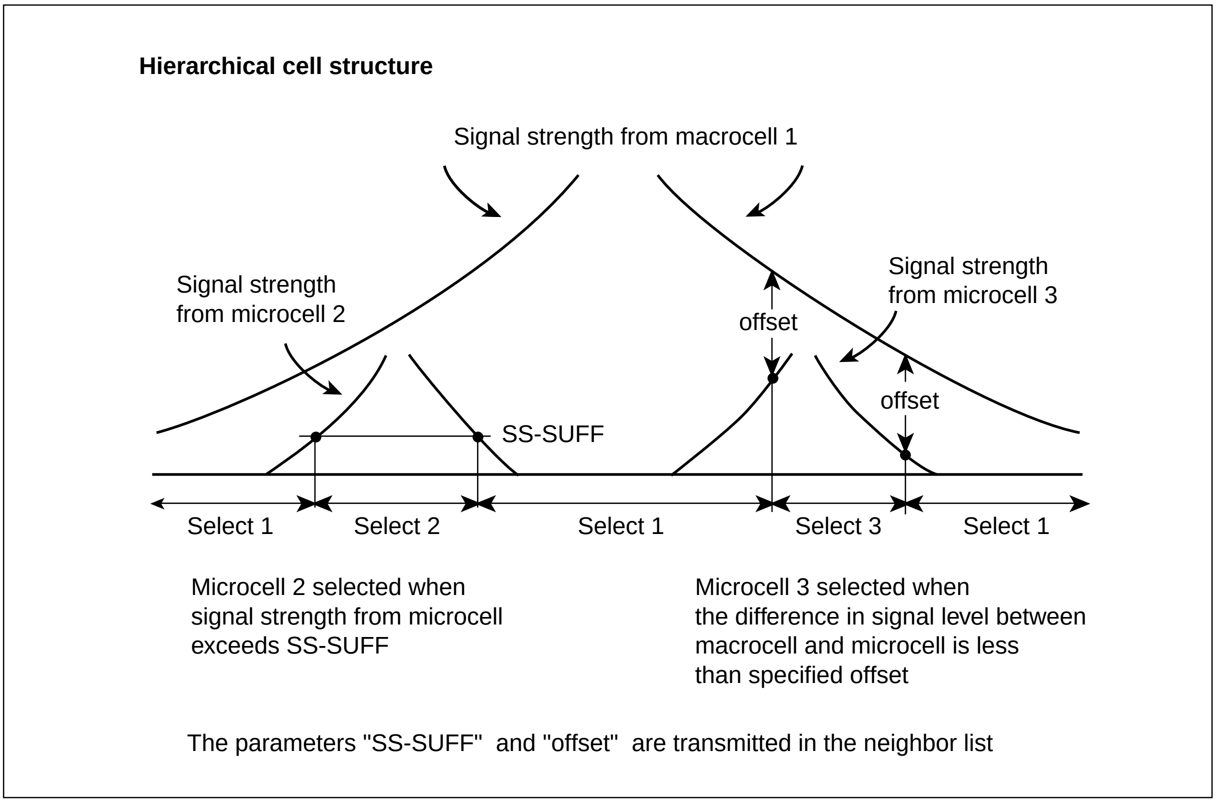


Figure 4

3.3 Virtual Mobile Location Area (VMLA)

Instead of fixed location areas used at GSM and the original AMPS, IS-54C utilizes a more flexible concept VMLA. When a mobile registers, the network control sends over a list of cell numbers, which defines the current location area for the mobile. When the mobile moves outside of the current location area and informs the network control about this, it is given a new list of cell numbers, defining a new location area. The advantages with this arrangement are:

• different classes of mobiles (i.e. with different speed characteristics) can be assigned location areas of different sizes

• the location area can be centered around the mobile.

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