# *Wireless Concepts*

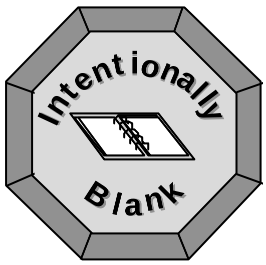
## *Chapter 3*

This chapter is designed to provide the student with an overview of basic concepts of wireless communications.

## OBJECTIVES:

Upon completion of this chapter the student will be able to:

- Briefly describe the Time Division Multiple Access technique (TDMA)
- List 2 transmission problems and their solutions
- Understand how Adaptive Multi-Rate (AMR) can increase capacity


Intentionally
Blank

# 3 Wireless Concepts

## Table of Contents

**Topic** **Page**

## FREQUENCY CONCEPTS

The following table summarizes the frequency-related specifications of each of the GSM systems. The terms used in the table are explained in the remainder of this section.

| System | GSM 800 | P-GSM 900 | E-GSM 900 | GSM 1800 | GSM 1900 |
|---|---|---|---|---|---|
| Frequencies:<br>• Uplink<br>• Downlink | <br>824-849 MHz<br>869-894 MHz | <br>890-915 MHz<br>935-960 MHz | <br>880-915 MHz<br>925-960 MHz | <br>1710-1785 MHz<br>1805-1880 MHz | <br>1850-1910 MHz<br>1930-1990 MHz |
| Wavelength | 37.5 cm | ~ 33 cm | ~ 33 cm | ~ 17 cm | ~ 16 cm |
| Bandwidth | 25 MHz | 25 MHz | 35 MHz | 75 MHz | 60 MHz |
| Duplex Distance | 45 MHz | 45 MHz | 45 MHz | 95 MHz | 80 MHz |
| Carrier Separation | 200 kHz | 200 kHz | 200 kHz | 200 kHz | 200 kHz |
| Radio Channels[1] | 125 | 125 | 175 | 375 | 300 |
| Transmission Rate | 270 kbits/s | 270 kbits/s | 270 kbits/s | 270 kbits/s | 270 kbits/s |

*Table 3-1 Frequency-related specifications*

[1] Note: Every GSM network uses one channel as a guard channel. This reduces the number of channels available for traffic by one. This is used to separate GSM frequencies from the frequencies of neighboring applications, e.g. 889 MHz. In this way extra protection (and quality) for GSM calls is ensured.

## FREQUENCY

> ☞ Did you know?
>
> *Due to frequency, a BTS transmitting information at 1800 MHz with an output power of 10 Watts (W) will cover only half the area of a similar BTS transmitting at 900 MHz. To counteract this, BTSs using 1800 MHz may use a higher output power.*

An MS communicates with a BTS by transmitting or receiving radio waves, which consist of electromagnetic energy. The frequency of a radio wave is the number of times the wave oscillates per second. Frequency is measured in Hertz (Hz), where 1 Hz indicates one oscillation per second. We also describe radio waves in terms of amplitude and phase. In simple terms amplitude is the voltage or height of the wave and phase is the form, or shape, of one oscillation over time.

Radio frequencies are used for many applications in the world today. Some common uses include:

- Television: 300 MHz approx.
- FM Radio: 100 MHz approx.
- Police radios: Country dependent
- Mobile networks: 300 - 2000 MHz approx.

The frequencies used by mobile networks vary according to the standard being used[2]. An operator applies for the available frequencies or, as in the United States, the operator bids for frequency bands at an auction. The following diagram displays the ?frequencies used by the major mobile standards:

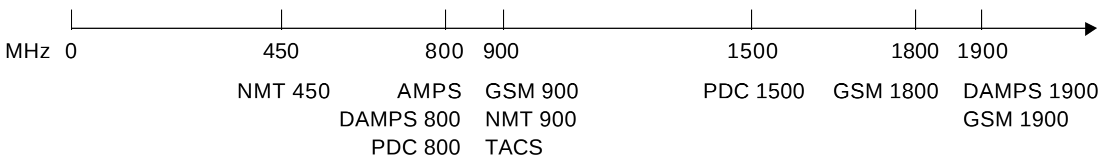



*Figure 3-1 Frequencies for major mobile standards*

[2] As these frequencies are used to carry information, they are often referred to as carrier frequencies.

## Wavelength

There are many different types of electromagnetic waves. These electromagnetic waves can be described by a sinusoidal function, which is characterized by wavelength. Wavelength (λ) is the length of one complete oscillation and is measured in meters (m). Frequency and wavelength are related via the speed of propagation, which for radio waves is the speed of light (3 x$10^8$ m/s or meters per second).

The wavelength of a frequency can be determined by using the following formula:

$$\text{Wavelength} = \frac{\text{Speed}}{\text{Frequency}}$$

Thus, for GSM 900 the wavelength is:

$$\text{Wavelength} = \frac{3 \times 10^8 \text{m/s}}{900 \text{ MHz}}$$

$$\text{Wavelength} = \frac{300{,}000{,}000 \text{ m/s}}{900{,}000{,}000}$$

Wavelength = 0.33 m (or 33 cm)

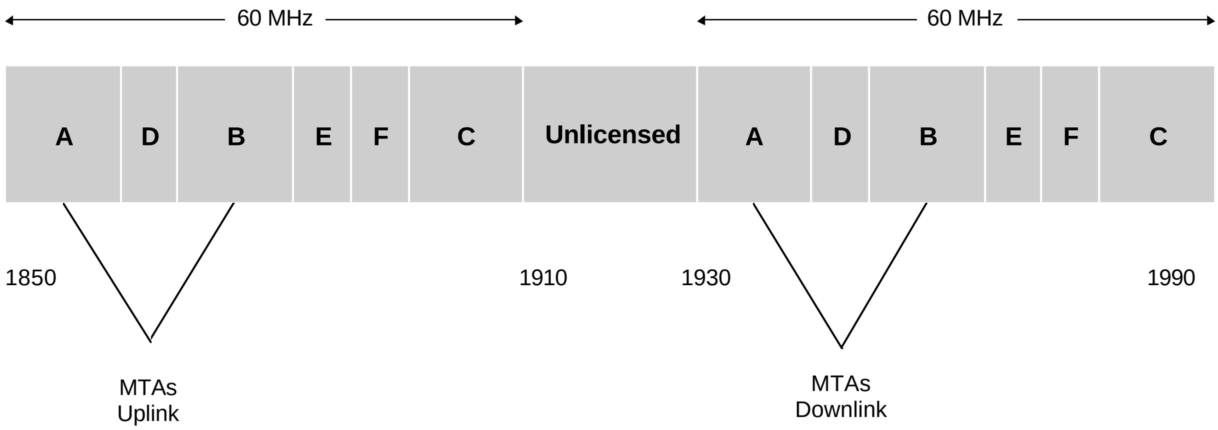
From this formula it can be determined that the higher the frequency, the shorter the wavelength. Lower frequencies, with longer wavelengths, are better suited to transmission over large distances, because they bounce on the surface of the earth and in the atmosphere. Television and FM radio are examples of applications, which use lower frequencies.

Higher frequencies, with shorter wavelengths, are better suited to transmission over small distances, because they are sensitive to such problems as obstacles in the line of the transmission path. Higher frequencies are suited to small areas of coverage, where the receiver is relatively close to the transmitter.

The frequencies used by mobile systems compromise between the coverage advantages offered by lower frequencies and the closeness-to-the-receiver advantages offered by use of higher frequencies.

## Example of Frequency Allocation - United States

In 1994, the Federal Communications Commission (FCC) in the United States (US) auctioned licenses to prospective mobile network operators. Each network operator owns the rights to the license for ten years. Further auctions will be held following that period for the same licenses. The FCC has specified six blocks within the frequency band: three duplex blocks A, B, and C (30 MHz each) and three other duplex blocks D, E, and F (10 MHz each).




| | | |
|---|---|---|
| A,B | 2 x 15 MHz | MTA |
| C | 2 x 15 MHz | BTA |
| D, E, F | 2 x 5 MHz | BTA |

*Figure 3-2 Spectrum allocation for PCS 1900 in the United States*

For telecommunications purposes, the US is divided into 51 regions or Major Trading Areas (MTA) and 493 Basic Trading Areas (BTA). One MTA can be as large in geographical area as a state, while a BTA can be about the size of a large city. The FCC issued two PCS 1900 licenses for each MTA and four licenses for each BTA[3]. Thus a city such as Los Angeles could be served for example by 6 operators: 2 MTA companies operating in California and 4 BTA companies operating in Los Angeles.

[3] Note: The choice of technology to use with the 1900 MHz frequencies is made by the operators. Both D-AMPS 1900 and GSM 1900 have been popular choices

## BANDWIDTH

Bandwidth is the term used to describe the amount of frequency range allocated to one application. The bandwidth given to an application depends on the amount of available frequency spectrum. The amount of bandwidth available is an important factor in determining the capacity of a mobile system, i.e. the number of calls, which can be handled.

## CHANNELS

Another important factor in determining the capacity of a mobile system is the channel. A channel is a frequency or set of frequencies which can be allocated for the transmission, and possibly the receipt, of information. Communication channels of any form can be one of the following types:

| Type | Description | Examples |
|---|---|---|
| Simplex | One way only | FM radio, television |
| Half duplex | Two way, only one at a time | Police radio |
| Full duplex | Two way, both at the same time | Mobile systems |

☞ Did you know?

*Because it requires less power to transmit a lower frequency over a given distance, uplink frequencies in mobile systems are always the lower band of frequencies - this saves valuable battery power of the MSs.*

A simplex channel, such as a FM radio music station, uses a single frequency in a single direction only. A duplex channel, such as that used during a mobile call, uses two frequencies: one to the MS and one from the MS. The direction from the MS to the network is referred to as *uplink*. The direction from the network to the MS is referred to as *downlink*.

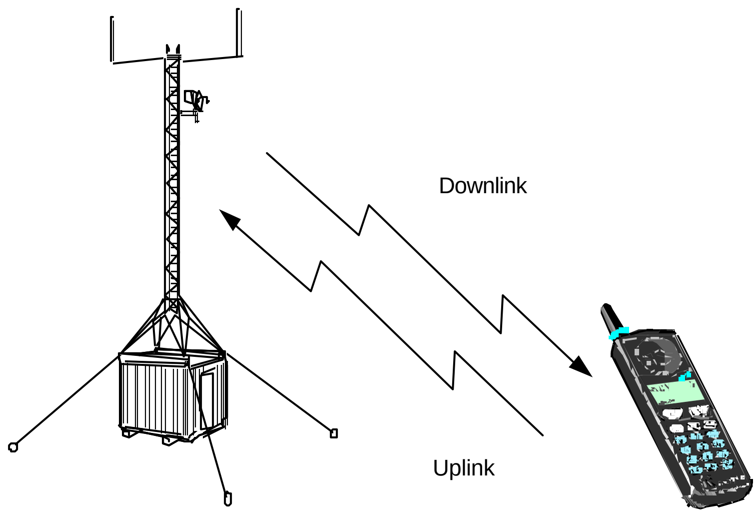



*Figure 3-3 Uplink and downlink on a radio channel*

## Duplex Distance

The use of full duplex requires that uplink and downlink transmissions are separated in frequency by a minimum distance, called duplex distance. Without it, uplink and downlink frequencies would interfere with each other.

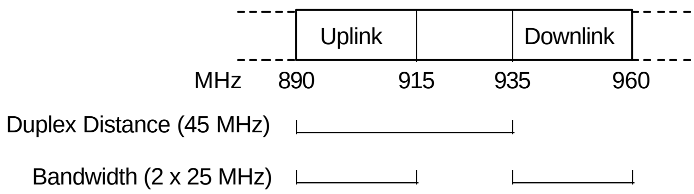



*Figure 3-4 Duplex Distance*

## Carrier Separation

In addition to the duplex distance, every mobile system includes a carrier separation[4]. This is the distance on the frequency band between channels being transmitted in the same direction. This is required in order to avoid the overlapping of information in one channel into an adjacent channel.

The length of separation between two channels is dependent on the amount of information to be transmitted within the channel. The greater the amount of information to transmit, the greater the amount of separation required. In GSM the carrier separation is fixed at 200 kHz

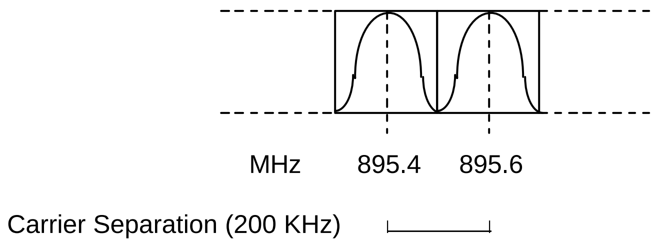



*Figure 3-5 Carrier separation*

From the figure above, it can be seen that the information to be sent is carried on the carrier frequency of 895.4 MHz. The same is true of the information to be sent on 895.6 MHz. To avoid interference between the two sets of information, a separation distance of 200 kHz is required. If less separation were used, they would interfere

[4] Carrier separation is sometimes referred to as carrier bandwidth.

and a caller on 895.4 MHz may experience crosstalk or noise from the caller on 895.6 MHz.

## Capacity and Frequency Re-use

It is the number of frequencies in a cell that determines the cell's capacity. Each company with a license to operate a mobile network is allocated a limited number of frequencies. These are distributed throughout the cells in their network. Depending on the traffic load and the availability of frequencies, a cell may have one or more frequencies allocated to it.

It is important when allocating frequencies that interference is avoided. Interference can be caused by a variety of factors. A common factor is the use of similar frequencies close to each other. The higher the interference, the lower the call quality.

To cover an entire country, for example, frequencies must be re-used many times at different geographical locations in order to provide a network with sufficient capacity. The same frequencies can not be re-used in neighboring cells as they would interfere with each other, so special patterns of frequency usage are determined during the planning of the network.

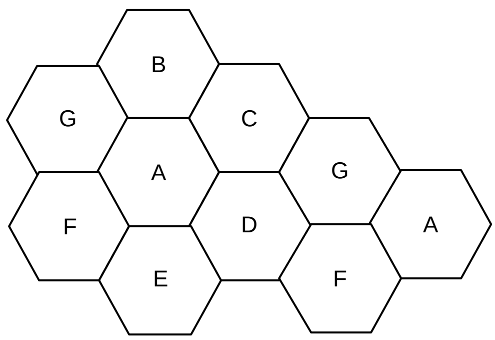



*Figure 3-6 Neighboring cells cannot have the same frequency (simplified)*

These frequency re-use patterns ensure that any frequencies being re-used are located at a sufficient distance apart to ensure that there is little interference between them. The term "frequency re-use distance" is used to describe the distance between two identical frequencies in a re-use pattern. The lower the frequency re-use distance, the more capacity will be available in the network.

## TRANSMISSION RATE

The amount of information transmitted over a radio channel over a period of time is known as the transmission rate. Transmission rate is expressed in bits per second or bit/s. In GSM the net bit rate over the air interface is 270kbit/s

## MODULATION METHOD

In GSM 900 a subscriber is allocated a timeslot, on a frequency, around 900 MHz. This is the frequency that will carry the voice or data, in digital format and so it is called a carrier frequency or, simply, a carrier. We shall examine shortly how voice is transformed from its original analog form into digital form, but, for now, let us look at how the carrier wave actually carries digital information.

At a basic level, for a carrier frequency to carry digital information we must be able to modify the carrier waveform in some way so that it represents digital one (1) and modify it again so that it represents digital zero (0). This modification process is called ‘modulation’ and there are different methods available. We can modify the amplitude, frequency or phase of the carrier so that it represents the bit pattern or digitized version of the input signal. For example, we can modify the amplitude of a waveform so that a slightly higher amplitude represents digital 1 and the input, or unmodified, waveform represents digital 0. Depending on the modulation method used, each modulation of the waveform can represent one or several bits.

Any modulation scheme increases the carrier load and hence is a limit on the capacity of the frequency band available. In GSM, the carrier bandwidth is 200 kHz.

The modulation technique used in GSM is Gaussian Minimum Shift Keying (GMSK) and is a form of phase modulation, or ‘phase shift keying’ as it is called.  GMSK enables the transmission of 270kbit/s within a 200kHz channel. This gives a bitrate of 1.3 bit/s per Hz. This is a rather low bitrate but acceptable as GMSK gives high interference resistance level.

The channel capacity in GSM does not compare favorably with other digital mobile standards, which can fit more bits/s onto a channel. In this way the capacity of other mobile standards is higher. However, GMSK offers more tolerance of interference. This in turn enables tighter re-use of frequencies and leads to an overall gain in capacity, which out-performs other systems.

## ACCESS METHOD: TIME DIVISION MULTIPLE ACCESS (TDMA)

> ☞ Did you know?
>
> *TDMA is not the only possible access method for mobile systems. Analog systems use Frequency DMA (FDMA) while some systems use Code DMA (CDMA).*

Most digital cellular systems use the technique of Time Division Multiple Access (TDMA) to transmit and receive speech signals. With TDMA, one carrier is used to carry a number of calls, each call using that carrier at designated periods in time. These periods of time are referred to as time slots. Each MS on a call is assigned one time slot on the uplink frequency and one on the downlink frequency. Information sent during one time slot is called a burst.

In GSM, a TDMA frame consists of 8 time slots. This means that a GSM radio carrier can carry 8 calls.

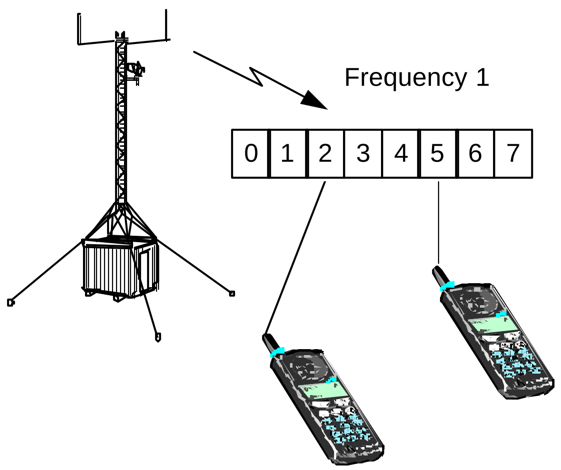



*Figure 3-7 TDMA*

Note: Only the downlink direction is shown. There is also a corresponding frame in the uplink direction.

# ANALOG AND DIGITAL TRANSMISSION

## INTRODUCTION TO ANALOG AND DIGITAL

### Analog Information

Analog information is continuous and does not stop at discrete values. An example of analog information is time. It is continuous and does not stop at specific points. An analog watch may have a second-hand, which does not jump from one second to the next, but continues around the watch face without stopping.

### Analog Signals

An analog signal is a continuous wave-form which changes in accordance with the properties of the information being represented.

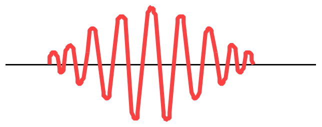


*Figure 3-8 Analog Signal*

### Digital Information

Digital information is a set of discrete values. Time can also be represented digitally. However, digital time would be represented by a watch which jumps from one minute to the next without stopping at the seconds. In effect, such a digital watch is taking a sample of time at predefined intervals.

### Digital Signals

For mobile systems, digital signals may be considered to be sets of discrete waveforms.

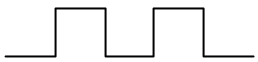


*Figure 3-9 Digital Signal*

## ADVANTAGES OF USING DIGITAL

Human speech is a form of analog information. It is continuous and changes in both frequency (higher and lower pitches) and amplitude (whispering and shouting).

At first, analog signals may appear to be a better medium for carrying analog information such as speech. Analog information is continuous and if it were to be represented by discrete samples of the information (digital signal), then some information would be missing (like the seconds on the digital watch). An analog signal would not miss any values, as it too is continuous.

All signals, analog and digital, become distorted over distances. In analog, the only solution to this is to amplify the signal. However, in doing so, the distortion is also amplified. In digital, the signal can be completely regenerated as new, without the distortion.

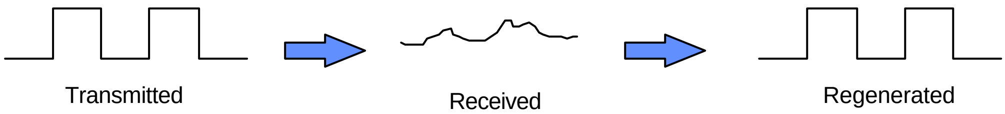



*Figure 3-10 Regeneration of digital signal*

The problem with using digital signals to transfer analog information is that some information will be missing due to the technique of taking samples. However, the more often the samples are taken, the closer the resulting digital values will be to a true representation of the analog information.

Overall, if samples are taken often enough, digital signals provide a better quality for transmission of analog information than analog signals.

# TRANSMISSION PROBLEMS

Many problems may occur during the transmission of a radio signal. Some of the most common problems are described below.

## PATH LOSS

Path loss occurs when the received signal becomes weaker and weaker due to increasing distance between MS and BTS, even if there are no obstacles between the transmitting (Tx) and receiving (Rx) antenna. The path loss problem seldom leads to a dropped call because before the problem becomes extreme, a new transmission path is established via another BTS.

## SHADOWING

Shadowing occurs when there are physical obstacles including hills and buildings between the BTS and the MS. The obstacles create a shadowing effect which can decrease the received signal strength. When the MS moves, the signal strength fluctuates depending on the obstacles between the MS and BTS.

A signal influenced by fading varies in signal strength. Drops in strength are called fading dips.

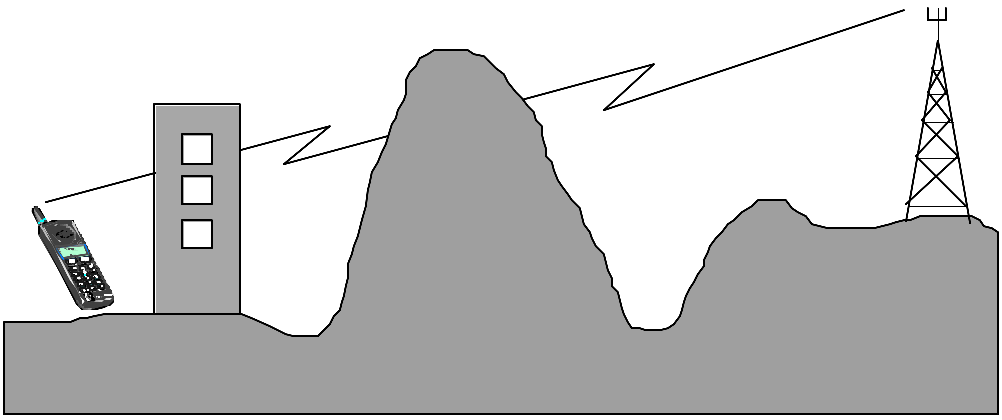


*Figure 3-11 Shadowing*

## MULTIPATH FADING

Multipath fading occurs when there is more than one transmission path to the MS or BTS, and therefore more than one signal is arriving at the receiver. This may be due to buildings or mountains, either close to or far from the receiving device.

Rayleigh fading and time dispersion are forms of multipath fading.

### Rayleigh fading

This occurs when a signal takes more than one path between the MS and BTS antennas. In this case, the signal is not received on a line of sight path directly from the Tx antenna. Rather, it is reflected off buildings, for example, and is received from several different indirect paths. Rayleigh fading occurs when the obstacles are close to the receiving antenna.

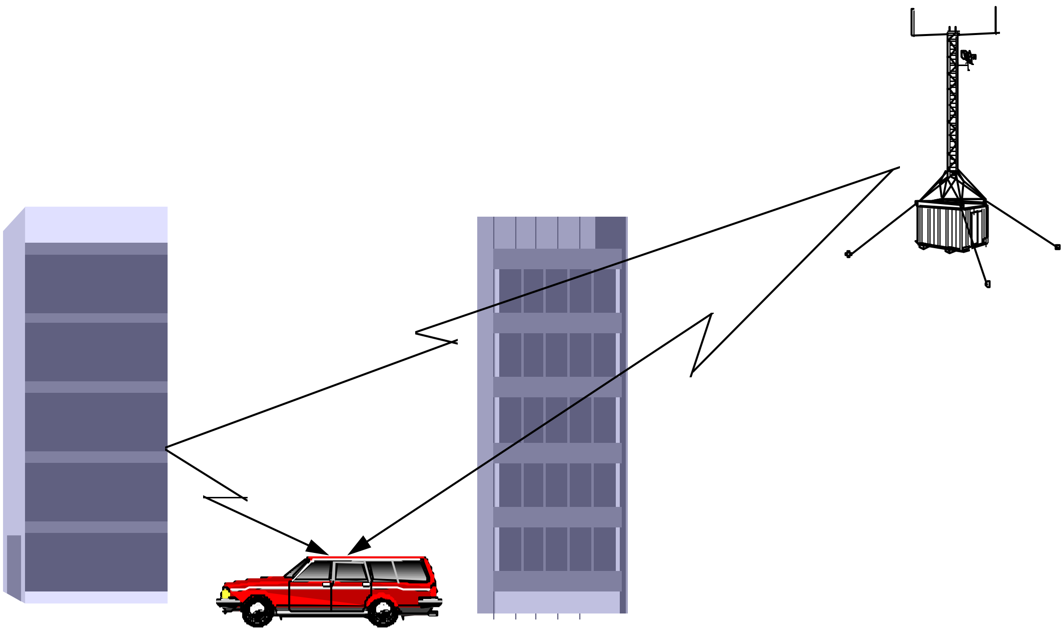


*Figure 3-12 Rayleigh fading*

The received signal is the sum of many identical signals that differ only in phase (and to some extent amplitude). A fading dip and the time that elapses between two fading dips depends on both the speed of the MS and the transmitting frequency. As an approximation, the distance between two dips caused by Rayleigh fading is about half a wavelength. Thus, for GSM 900 the distance between dips is about 17 cm.

## Time Dispersion

Time dispersion is another problem relating to multiple paths to the Rx antenna of either an MS or BTS. However, in contrast to Rayleigh fading, the reflected signal comes from an object far away from the Rx antenna.

Time dispersion causes Inter-Symbol Interference (ISI) where consecutive symbols (bits) interfere with each other making it difficult for the receiver to determine which symbol is the correct one. An example of this is shown in the figure below where the sequence 1, 0 is sent from the BTS.

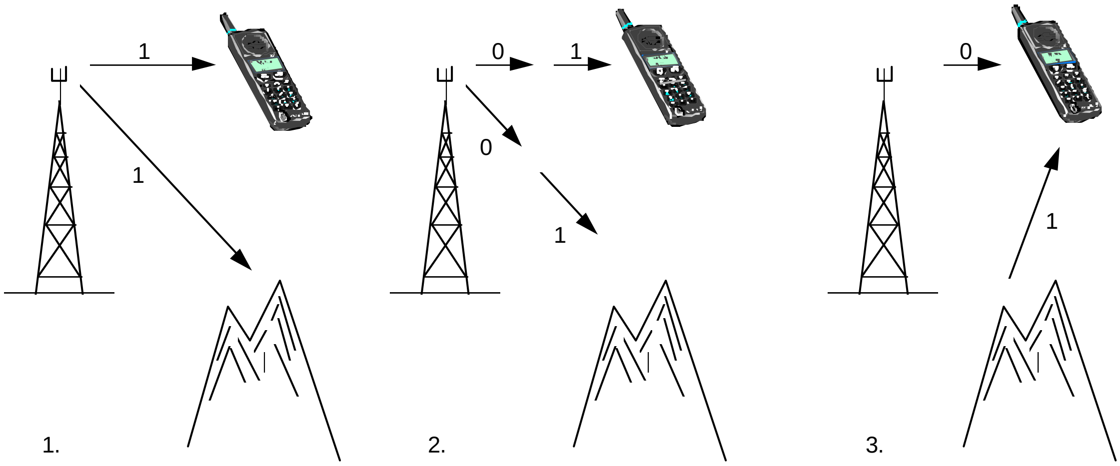



*Figure 3-13 Time dispersion*

☞ Did you know?

*One bit is transmitted every 3.7 µs. Radio waves travel at $3x10^8$ m/s. Therefore, one bit travels approximately 1 km within one bit period.*

*Thus, if the direct path is 1km and the indirect path is 3 km long, the first bit transmitted will interfere with the 3rd bit transmitted.*

If the reflected signal arrives one bit time after the direct signal, then the receiver detects a 1 from the reflected wave at the same time it detects a 0 from the direct wave. The symbol 1 interferes with the symbol 0 and the MS does not know which one is correct.

## TIME ALIGNMENT

Each MS on a call is allocated a time slot on a TDMA frame. This is an amount of time during which the MS transmits information to the BTS. The information must also arrive at the BTS within that time slot. The time alignment problem occurs when part of the information transmitted by an MS does not arrive within the allocated time slot. Instead, that part may arrive during the next time slot, and may interfere with information from another MS using that other time slot.

A large distance between the MS and the BTS causes time alignment. Effectively, the signal cannot travel over the large distance within the given time.

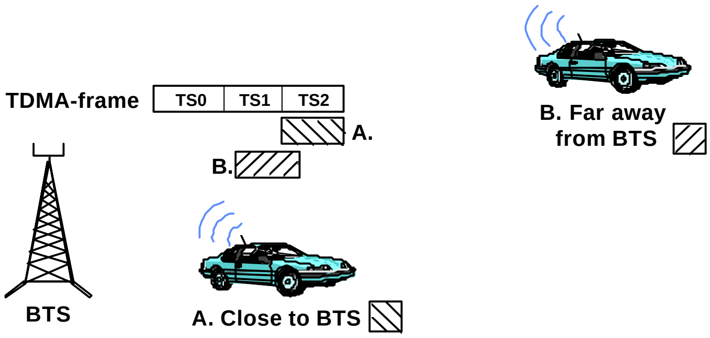



*Figure 3-14 The time alignment problem*

For example, an MS is close to a BTS and has been allocated time slot 3 (TS 3). During the call, the MS moves away from the BTS causing the information sent from the BTS to arrive at the MS later and later. The answer from the MS also arrives later at the BTS. If nothing is done, the delay becomes so long that the transmission from the MS in time slot 3 overlaps with the information which the BTS receives in time slot 4.

## COMBINED SIGNAL LOSS

Each of the problems described above occurs independently of each other. However, in most calls some of these problems may occur at the same time. An illustration of what the signal strength may look like at the MS Rx antenna when moving away from the BTS Tx antenna is shown in Figure 3-15. The problems of path loss, shadowing and Rayleigh fading are present for this transmission path.

The signal strength as a global mean value decreases with the distance (path loss) and finally results in a lost connection. Around this global mean, slow variations are present due to shadowing effects and fast variations are present due to Rayleigh fading.

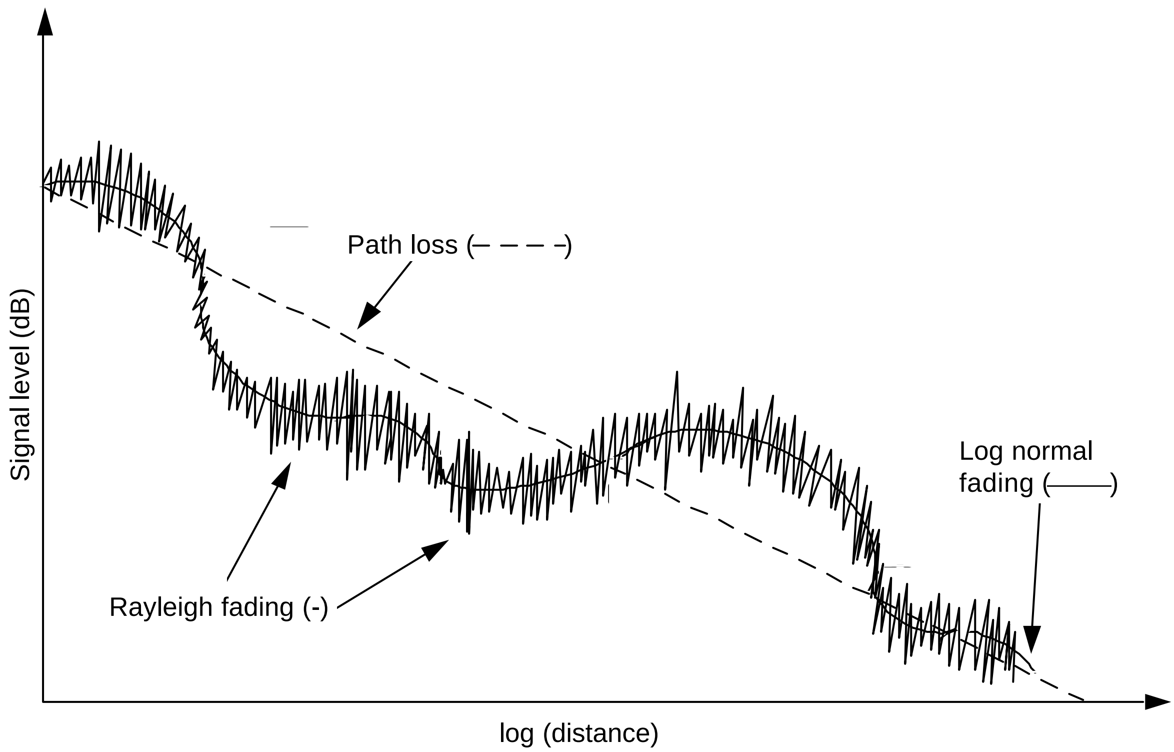



*Figure 3-15 Rx signal strength versus distance*

At any one point from the Tx antenna, the received signal can look like the signal in Figure 3-16 below.

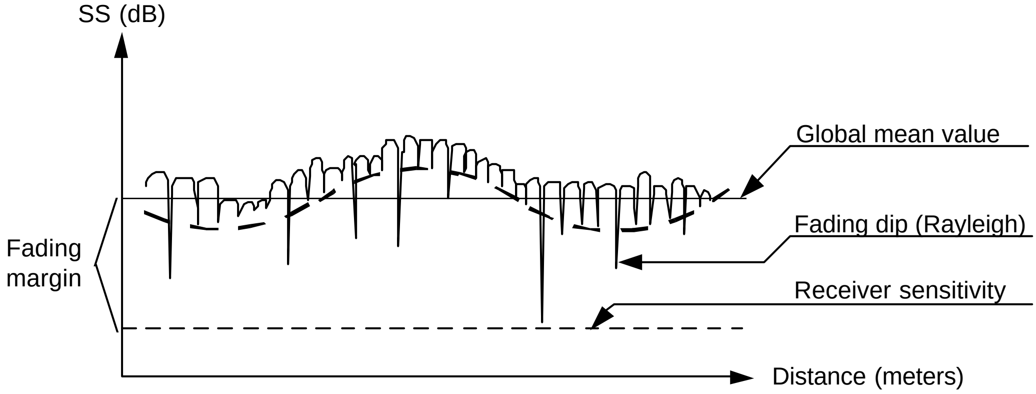



*Figure 3-16 Rx signal strength*

The lowest signal strength value required for a specified output is called receiver sensitivity level. To detect the information sent from Tx antenna, X watts must be received. If the signal falls below X, the information will be lost and the call may be dropped. To ensure that no information is lost, the global mean value must be as many dB above the receiver sensitivity level as the strongest (deepest) fading dip gives rise to. This fading margin is the difference between the global mean value and the receiver sensitivity.

# SOLUTIONS TO TRANSMISSION PROBLEMS

This section describes some solutions to the problems described in previous sections. Although many of these do not entirely solve all problems on the radio transmission path, they do play an important part in maintaining call quality for as long as possible.

## CHANNEL CODING

In digital transmission, the quality of the transmitted signal is often expressed in terms of *how many of the received bits are incorrect*. This is called Bit Error Rate (BER). BER defines the percentage of the total number of received bits which are incorrectly detected.

| | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| **Transmitted bits** | 1 | 1 | 0 | 1 | 0 | 0 | 0 | 1 | 1 | 0 | |
| **Received bits** | 1 | 0 | 0 | 1 | 0 | 0 | 1 | 0 | 1 | 0 | |
| **Errors** | | ↑ | | | | | ↑ | ↑ | | | 3/10 = 30% BER |

This percentage should be as low as possible. It is not possible to reduce the percentage to zero because the transmission path is constantly changing. This means that there must be an allowance for a certain amount of errors and at the same time an ability to restore the information, or at least detect errors so the incorrect information bits are not interpreted as correct. This is especially important during transmission of data, as opposed to speech, for which a higher BER is acceptable.

Channel coding is used to detect and correct errors in a received bit stream. It adds bits to a message. These bits enable a channel decoder to determine whether the message has faulty bits, and to potentially correct the faulty bits.

## ADAPTIVE MULTI RATE (AMR)

We know now that channel coding provides a way of protecting digital information over the air interface. The amount of channel coding used depends on how the GSM network is configured.

As we shall see when we examine the GSM transmission process, the standard GSM configuration inserts a fixed number of channel coding bits per TDMA timeslot. If however the network is equipped with Adaptive Multi Rate (AMR), then the rate of channel coding bits and the underlying speech codec rate can be adapted to suit the prevailing radio environment.

AMR consists of a number of different codecs, which together with the associated channel coding has been optimized for different radio environments. Depending on the measured Channel Interference Ratio (C/I) conditions, the best speech codec rate for the present conditions is chosen, which results in a significant improvement in speech quality. The possibility to increase/decrease the amount of channel coding depending on the C/I makes the channel more robust to bit errors. This more robust channel coding makes it possible to tighten the frequency planning and by that, increase the capacity in the radio network.

There are, in total, 8 speech codecs defined for AMR of which 6 have been defined for use in half rate (HR) channels. The difference between when a codec is used in a full rate (FR) channel and a half rate channel is the amount of channel coding, which is much more in a full rate channel. In short, FR channels provide better protection over the air interface and therefore better speech quality.

The 8 speech codec rates in AMR are:

12.2 kbits/s
10.2 kbits/s
7.95 kbits/s
7.40 kbits/s
6.70 kbits/s
5.90 kbits/s
5.15 kbits/s
4.75 kbits/s

For each mode, FR or HR codec sets are predefined (presently 2 sets per mode). Each coding set consists of up to four of the above codec rates. For each call, one coding set is selected. This coding set is called the Active Coding Set.

For each of the predefined coding sets there is an associated set of decision thresholds that determine which codec rate should be used for a certain C/I value. It is possible to change the codec rate every second speech frame but only to the next higher or lower codec rate in the active coding set. (Note: It is possible to switch between FR and HR with intra-cell handover.)

It is the receiving side (MS and BTS) that performs quality measurements on the incoming link to perform the codec rate adaptation. The codec rate changes are not audible.

The same AMR codec has also been specified for use in WCDMA/UMTS networks. This will guarantee similar speech quality in both GSM and UMTS.

## INTERLEAVING

In reality, bit errors often occur in sequence, as caused by long fading dips affecting several consecutive bits. Channel coding is most effective in detecting and correcting single errors and short error sequences. It is not suitable for handling longer sequences of bit errors.

> ☞ Did you know?
>
> *Interleaving could be compared to sending a group of important people from A to B on different planes. By doing so, the likelihood of losing the entire group is minimised.*

For this reason, a process called interleaving is used to separate consecutive bits of a message so that these are transmitted in a non-consecutive way.

For example, a message block may consist of four bits (1234). If four message blocks must be transmitted, and one is lost in transmission, without interleaving there is a 25% BER overall, but a 100% BER for that lost message block. It is not possible to recover from this.

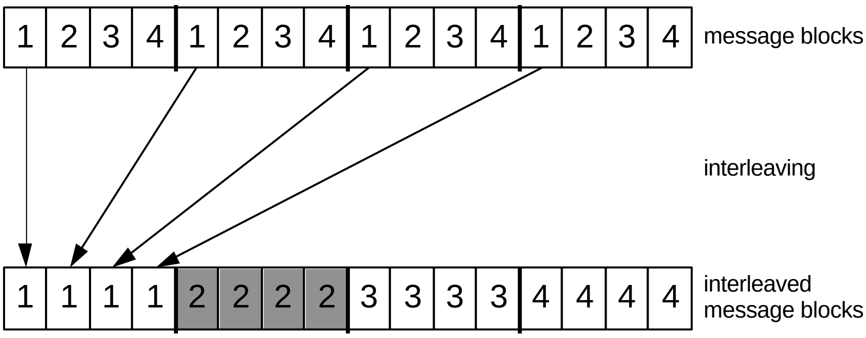



*Figure 3-17 Interleaving*

If interleaving is used, as shown in Figure 3-18, the bits of each block may be sent in a non-consecutive manner. If one block is lost in transmission, again there is a 25% BER overall. However, this time the 25% is spread over the entire set of message blocks, giving a 25% BER for each. This is more manageable and there is a greater possibility that the channel decoder can correct the errors.

| | | | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| 1 | X | 3 | 4 | 1 | X | 3 | 4 | 1 | X | 3 | 4 | 1 | X | 3 | 4 |

*Figure 3-18 Received interleaved message blocks*

# ANTENNA DIVERSITY

Antenna diversity increases the received signal strength by taking advantage of the natural properties of radio waves. There are two primary diverstiy methods: space diversity and polarization diversity.

## Space Diversity

An increased received signal strength at the BTS may be achieved by mounting two receiver antennae instead of one. If the two Rx antennae are physically separated, the probability that both of them are affected by a deep fading dip at the same time is low. At 900 MHz, it is possible to gain about 3 dB with a distance of five to six meters between the antennae. At 1800 MHz the distance can be shortened because of its decreased wavelength.

By choosing the best of each signal, the impact of fading can be reduced. Space diversity offers slightly better antenna gain than polarization diversity, but requires more space.

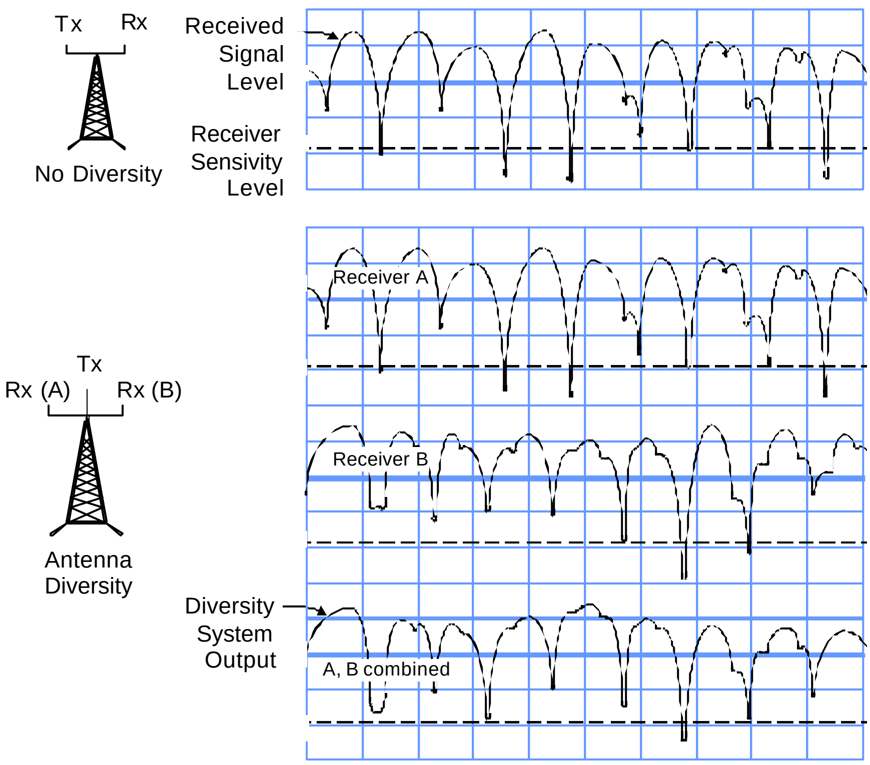



*Figure 3-19 Space diversity*

## Polarization Diversity

With polarisation diversity the two space diversity antennae are replaced by one dual polarized antenna. This antenna has normal size but contains two differently polarized antenna arrays. The most common types are vertical/horizontal arrays and arrays in ±45 degree slant orientation. The two arrays are connected to the respective Rx branches in the BTS. The two arrays can also be used as combined Tx/Rx antennas. For most applications, the difference between the diversity gain for space diversity and polarization diversity is negligible, but polarization diversity reduces the space required for antennae.

# ADAPTIVE EQUALIZATION

Adaptive equalization is a solution specifically designed to counteract the problem of time dispersion. It works as follows:

1. Eight sets of predefined known bit patterns exist, known as training sequences. These are known to the BTS and the MS (programmed at manufacture). The BTS instructs the MS to include one of these in its transmissions to the BTS.

2. The MS and BTS includes the training sequence (shown in the figure as “S”) in its transmissions.

3. The other party receives the transmission and examines the training sequence within it. The received training sequence is compared with the known training sequence that is used in this cell. It can be assumed that problems in the radio path affected these bits must also have had a similar affect on the speech data bits sent in the same burst.

4. The receiver begins a process in which it uses its knowledge of what happened the training sequence to correct the speech data bits of the transmission.

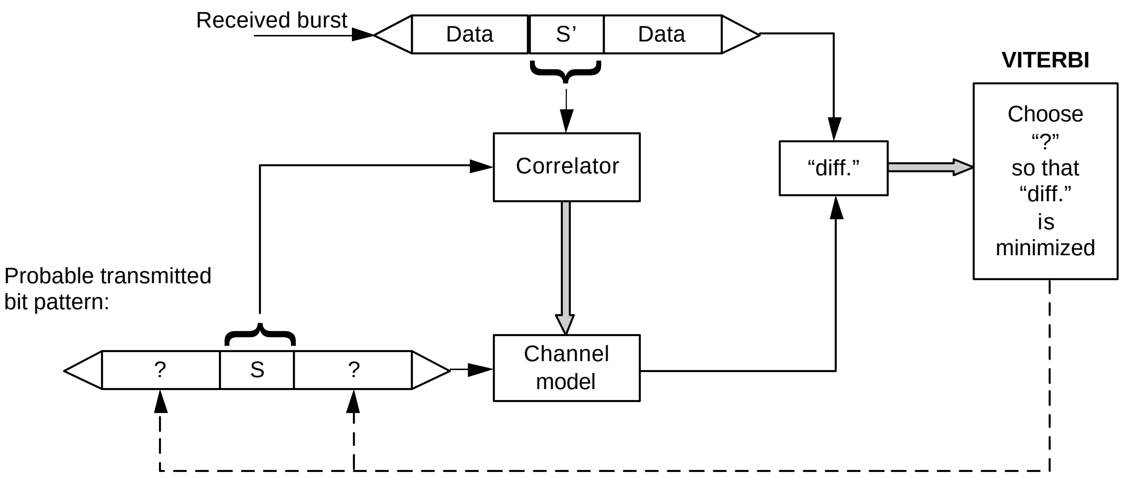



*Figure 3-20 Adaptive equalization*

Because some assumptions are made about the radio path, adaptive equalization may not result in a 100% perfect solution every time. However, a “good enough” result will be achieved. A viterbi equalizer is an example of an adaptive equalizer.

## FREQUENCY HOPPING

As mentioned previously, Rayleigh fading is frequency dependent. This means that the fading dips occur at different places for different frequencies. To benefit from this fact, it is possible for the BTS and MS to hop from frequency to frequency during a call. The frequency hopping of the BTS and MS is synchronized.

In GSM there are 64 patterns of frequency hopping, one of them is a simple cyclic or sequential pattern. The remaining 63 are known as pseudo-random patterns, which an operator can choose from.

Two types of hopping are supported by the BSC:

- Baseband hopping involves hopping between frequencies on different tranceivers in a cell.

- Synthesizer hopping involves hopping from frequency to frequency on the same tranceiver in a cell.

It is now possible to assign up to 32 frequencies to a channel group when synthesized frequency hopping is used, as compared to 16 previously. In many networks there are around 20 frequencies available for frequency hopping in a cell. In these networks 32 frequencies per channel group will be an advantage as it will now no longer be necessary to split the 20 frequencies between two channel groups.

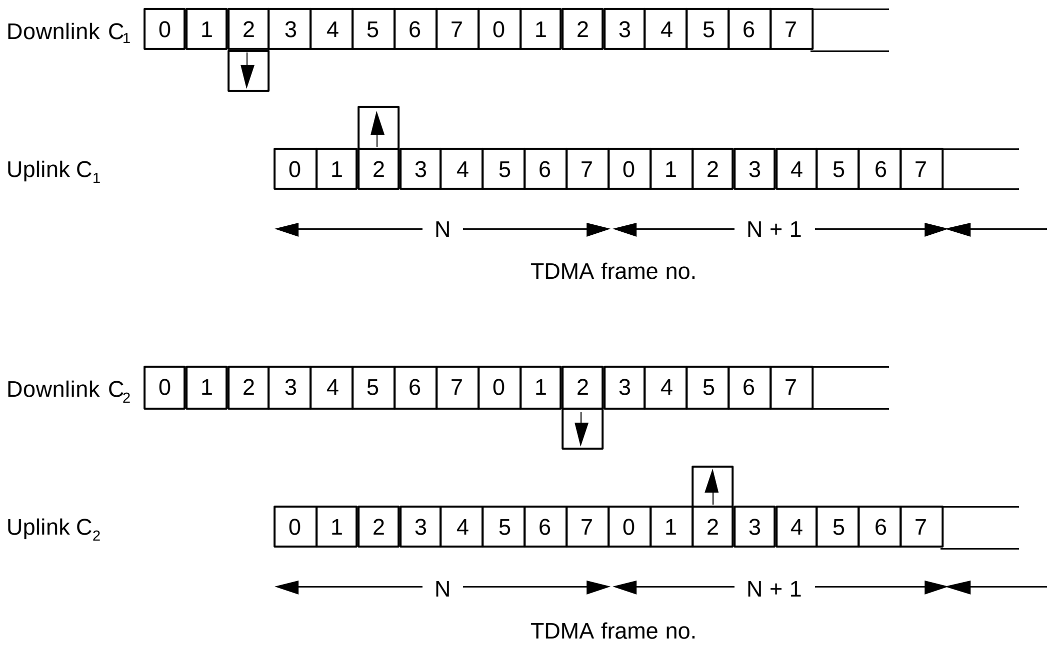


*Figure 3-21 Frequency hopping*

During TDMA frame N, $C_1$ is used and during TDMA frame N+1, $C_2$ is used. The call uses the same time slot but changes frequencies according to an identified pattern.

## TIMING ADVANCE

Timing advance is a solution specifically designed to counteract the problem of time alignment. It works by instructing the mis-aligned MS to transmit its burst earlier or later than it normally would.

In GSM, the timing advance information relates to bittimes. Thus, an MS may be instructed to commence its transmission a certain number of bittimes earlier or later, related to previous position, to reach its timeslot at the BTS in right time. Maximum 63 bittimes can be used in standard GSM systems. This limits GSM normal cell size to 35km radius. However with extended range equipment, distances up to 70km or even 121km can be handled, using 2 timeslots and both speech and single slot GPRS are supported. Because 2 timeslots are required this results in a drop in the number of available channels in the cell by 50%.

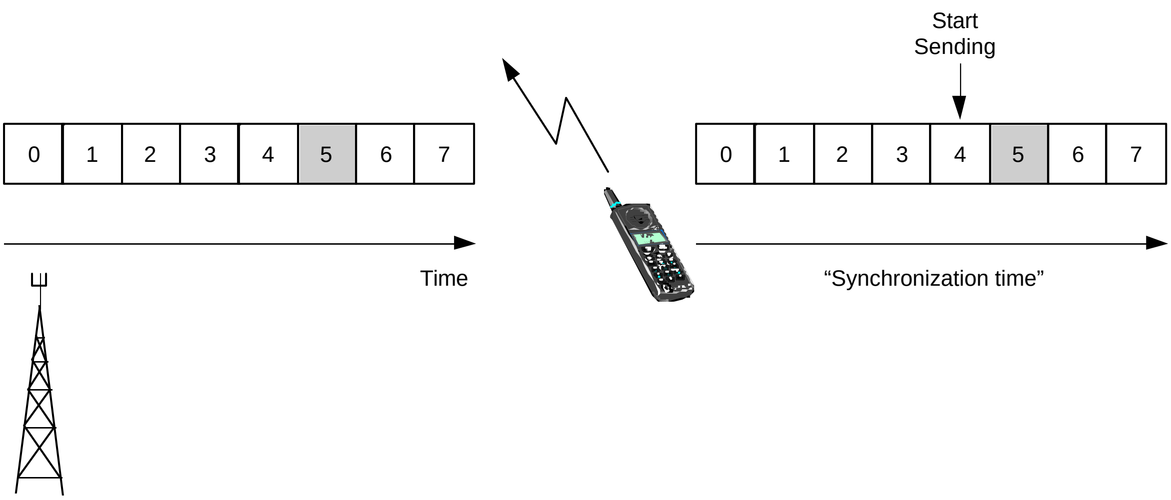



*Figure 3-22 Timing advance*

# GSM TRANSMISSION PROCESS

## STAGE 1: ANALOG TO DIGITAL (A/D) CONVERSION

One of the primary functions of an MS is to convert the analog speech information into digital form for transmission using a digital signal. The analog to digital (A/D) conversion process outputs a collection of bits: binary ones and zeros which represent the speech input.

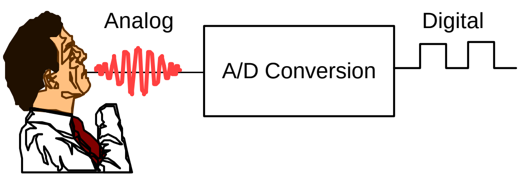



*Figure 3-23 A/D conversion*

The A/D conversion is performed by using a process called Pulse Code Modulation (PCM). PCM involves three main steps:

- Sampling
- Quantization
- Coding

### Step 1: Sampling

Sampling involves measuring the analog signal at specific time intervals.

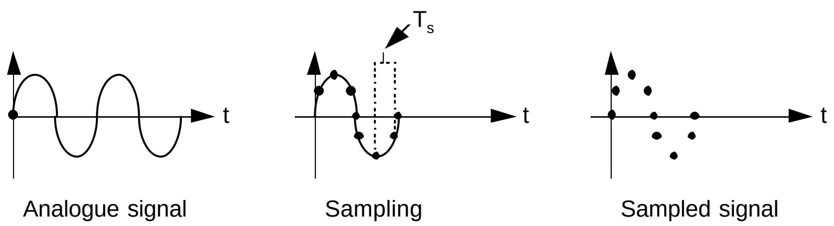



*Figure 3-24 Analog signal sampling*

The accuracy of describing the analog signal in digital terms depends on how often the analog signal is sampled. This is expressed as the sampling frequency. The sampling theory states that:

*To reproduce an analog signal without distortion, the signal must be sampled with at least twice the frequency of the highest frequency component in the analog signal.*

Normal speech mainly contains frequency components lower than 3400 Hz. Higher components have low energy and may be omitted without affecting the speech quality much. Applying the sampling theory to analog speech signals, the sampling frequency, should be at least 2 x 3.4 kHz = 6.8 kHz. Telecommunication systems use a sampling frequency of 8 kHz, which is acceptable based on the sampling theory.

## Step 2: Quantization

The next step is to give each sample a value. For this reason, the amplitude of the signal at the time of sampling is measured and approximated to one of a finite set of values. The figure below shows the principle of quantization applied to an analog signal. It can be seen that a slight error is introduced in this process when the signal is quantized or approximated. The degree of accuracy depends on the number of quantization levels used. Within common telephony, 256 levels are used while in GSM 8,192 levels are used.

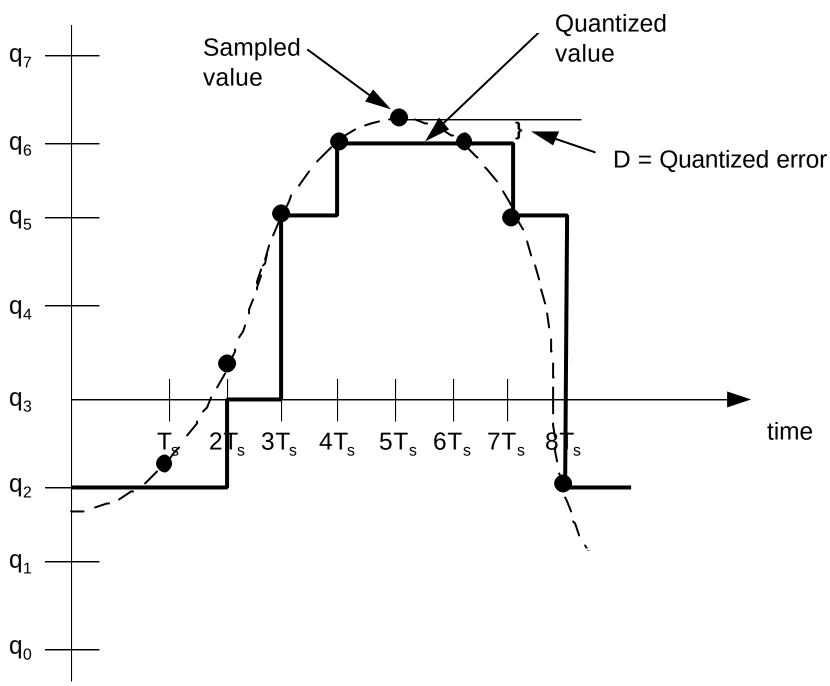



*Figure 3-25 Quantization*

## Step 3: Coding

Coding involves converting the quantized values into binary. Every value is represented by a binary code of 13 bits ($2^{13} = 8192$). For example, a quantized value of 2,157 would have a bit pattern of 0100001101101:

| **Bit** | **12** | **11** | **10** | **9** | **8** | **7** | **6** | **5** | **4** | **3** | **2** | **1** | **0** | **Total** |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| **Set to** | 0 | 1 | 0 | 0 | 0 | 0 | 1 | 1 | 0 | 1 | 1 | 0 | 1 | |
| **Value** | 0 | 2048 | 0 | 0 | 0 | 0 | 64 | 32 | 0 | 8 | 4 | 0 | 1 | **2157** |

*Table 3-2 Coding of quantised value 2157*

## Summary of A/D Conversion

The result from the process of A/D conversion is 8,000 samples per second of 13 bits each. This is a bit rate of 104 kbits/s.

When it is considered that 8 subscribers use one radio channel, the overall bit rate would be 8 x 104 kbits/s = 832 kbits/s. Recalling the general rule of 1 bit per Hertz, this bit rate would not fit into the 200 kHz available for all 8 subscribers. The bit rate must be reduced somehow - this is achieved using segmentation and speech coding.

## STAGE 2: SEGMENTATION AND STAGE 3: SPEECH CODING

☞ Did you know?

*In his childhood, Alexander Graham Bell constructed an artificial speaking machine. It was an anatomical model of the human voice tract, complete with teeth, throat, nasal passages and tongue.*

*By carefully positioning these elements, while simultaneously introducing a sound source in the throat, Bell was able to articulate simple English words.*

The key to reducing the bit rate is to send information about the speech instead of the speech itself. This can be explained with the following analogy:

> Person A wishes to listen to a certain piece of music and they know that person B has it on record. A rings B asking for the use of the record for some time. Unfortunately, the record is scratched and cannot be used. Instead, B sends A parameters of how the music is built up - the sheets of music - together with information about how fast it should be played - the frequency - and A reproduces the music.

In GSM, the speech coding process analyzes speech samples and outputs parameters of what the speech consists of the tone, length of tone, pitch, etc. This is then transmitted through the network to another MS, which generates the speech based on these parameters.

The process of segmentation and speech coding is explained in more detail as follows:

The human speech process starts in the vocal chords or speech organs, where a tone is generated. The mouth, tongue, teeth, etc. act as a filter, changing the nature of this tone. The aim of speech coding in GSM is to send only information about the original tone itself and about the filter.

Segmentation: Given that the speech organs are relatively slow in adapting to changes, the filter parameters representing the speech organs are approximately constant during 20 ms. For this reason, when coding speech in GSM, a block of 20 ms is coded into one set of bits. In effect, it is similar to sampling speech at a rate of 50 times per second instead of the 8,000 used by A/D conversion.

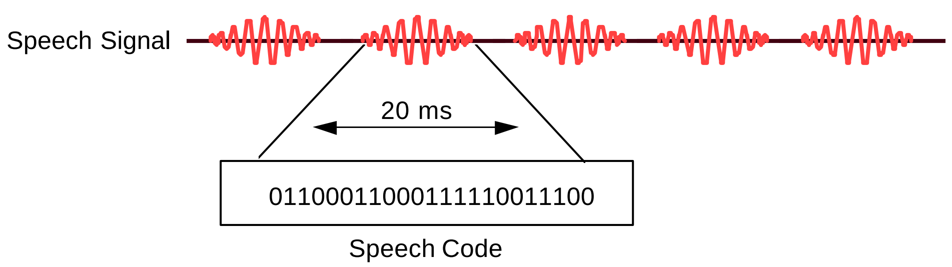



*Figure 3-26 Segmentation and speech coding*

Speech Coding[5]: Instead of using 13 bits per sample as in A/D conversion, GSM speech coding uses 260 bits. This calculates as 50 x 260 = 13 kbits/s. This provides a speech quality which is acceptable for mobile telephony and comparable with wireline PSTN phones.

Many types of speech coders are available. Some offer better speech quality, at the expense of a higher bit rate (waveform coders). Others use lower bit rates, at the expense of lower speech quality (vocoders). The hybrid coder which GSM uses provides good speech quality with a relatively low bit rate, at the expense of speech coder complexity.

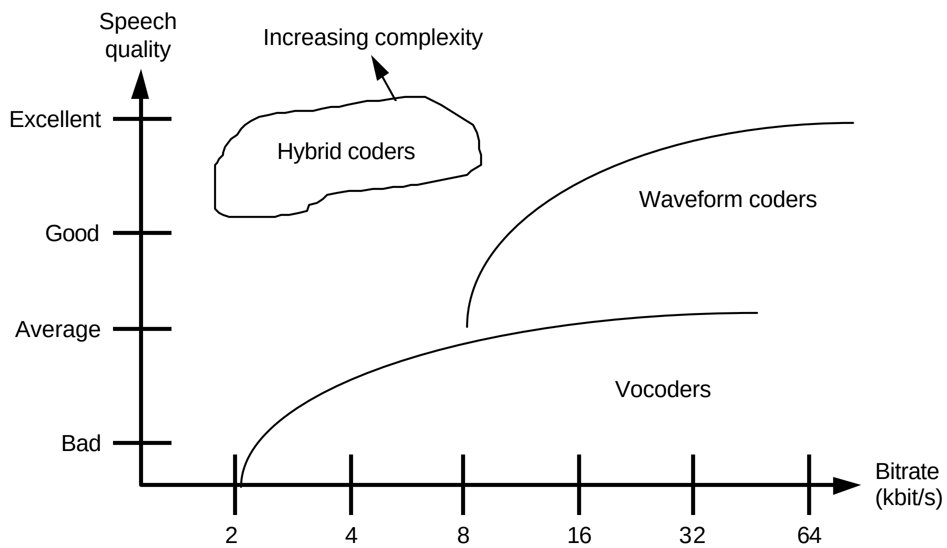



*Figure 3-27 Speech quality vs. bit rate*

## Summary of Segmentation and Speech Coding

> ☞ Did you know?
>
> *The speech coding process described here is for full rate speech only. Alternatives are:*
>
> - *Half rate: 6.5 kbits/s*
> - *Enhanced Full Rate (EFR): 13.0 kbits/s*

The GSM speech coder produces a bit rate of 13 kbits/s per subscriber. When it is considered that 8 subscribers use one radio channel, the overall bit rate would be 8 x 13 kbits/s = 104 kbits/s. This compares favorably with the 832 kbits/s from A/D conversion.

However, speech coding does not consider the problems which may be encountered on the radio transmission path. The next stages in the transmission process, channel coding and interleaving, help to overcome these problems.

[5] The function of converting from PCM coded information to GSM speech coder information is called transcoding.

## STAGE 4: CHANNEL CODING

Channel coding in GSM uses the 260 bits from speech coding as input to channel coding and outputs 456 encoded bits.

The 260 bits are split according to their relative importance:

- Block 1: 50 very important bits
- Block 2: 132 important bits and
- Block 3: 78 not so important bits

The first block of 50 bits is sent through a block coder, which adds three parity bits that will result in 53 bits. These three bits are used to *detect errors* in a received message.

The 53 bits from first block, the 132 bits from the second block and 4 tail bits (total = 189) are sent to a 1:2 convolutional coder which outputs 378 bits. Bits added by the convolutional coder *enable the correction of errors* when the message is received.

The bits of block 3 are not protected.

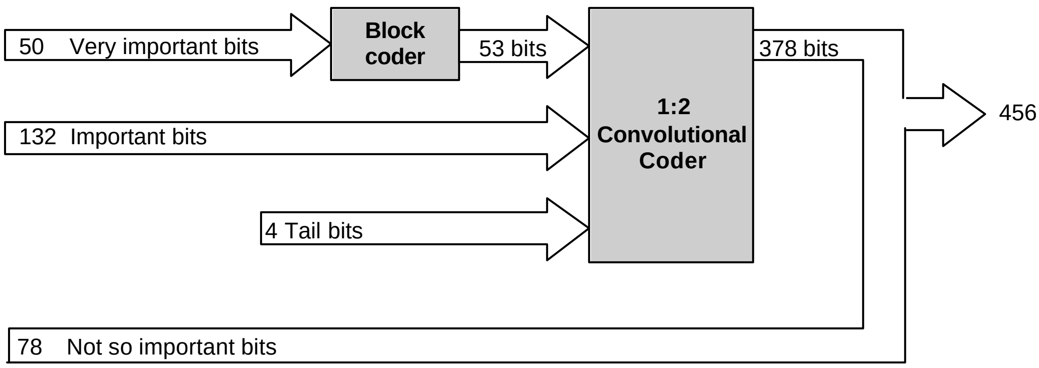



*Figure 3-28 Channel coding*

## *STAGE 5: INTERLEAVING*

## First level of interleaving

The channel coder provides 456 bits for every 20 ms of speech which are interleaved in eight blocks of 57 bits shown below.

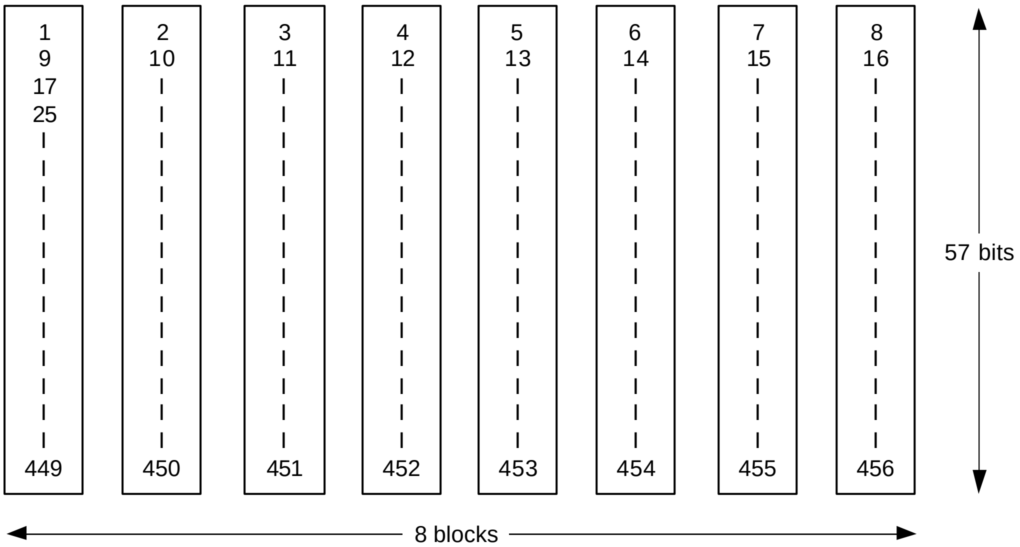



*Figure 3-29 Interleaving of 20 ms of encoded speech*

In a normal burst there is space for two of these speech blocks (Figure 3-30). (Remaining bits are explained later in this book.) Thus, if one burst transmission is lost, there is a 25% BER for the entire 20 ms of speech (2/8 = 25%).

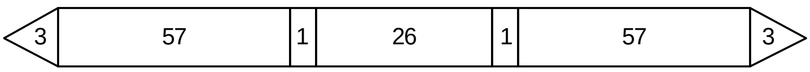



*Figure 3-30 Normal burst*

## Second level of interleaving

If only one level of interleaving is used, a loss of this burst results in a total loss of 25%. This is too much for the channel decoder to correct. A second level of interleaving can be introduced to further reduce the possible BER to 12.5%.

Instead of sending two blocks of 57 bits from the same 20 ms of speech within one burst, a block from one 20 ms and a block from next sample of 20 ms are sent together. A delay is introduced in the system when the MS must wait for the next 20 ms of speech. However, the system can now afford to loose a whole burst, out of eight, as the loss is only 12.5% of the total bits from each 20ms speech frame. 12,5% is the maximum loss level that channel decoder can correct.

| A | B | C | D |
|---|---|---|---|
| 20 ms speech<br>456 bits = 8x57 | **20 ms speech<br>456 bits = 8x57** | 20 ms speech<br>456 bits = 8x57 | 20 ms speech<br>456 bits = 8x57 |

*Figure 3-31 Speech frame*

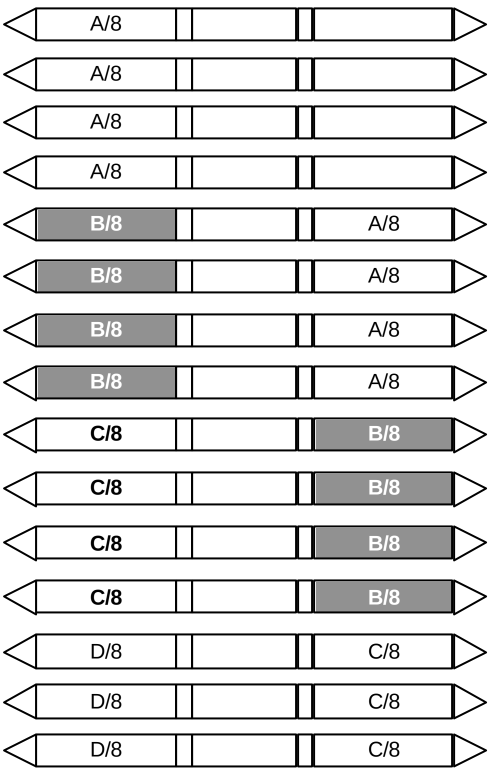



*Figure 3-32 Second level of interleaving*

## STAGE 6: CIPHERING/ENCRYPTION

The purpose of ciphering is to encode the burst so that it cannot be interpreted by any other device than the intended receiver. The ciphering algorithm in GSM is called the A5 algorithm. It does not add bits to the burst, meaning that the input and output to the ciphering process is the same as the input: 456 bits per 20 ms.

## STAGE 7: BURST FORMATTING

As previously explained, every transmission from an MS/BTS must include some extra information such as the 26 training sequence bits, 2 flag bits and 6 tail bits. The process of burst formatting is to add these bits to the basic speech/data (57+57=114 bits) being sent. Consequently this increases the  burst from 114 to 148 bits, thus increasing the transmission rate on the air, but is necessary to counteract problems encountered on the radio path.

In GSM, the input to burst formatting for 20ms speech is the 456 bits received from ciphering. Burst formatting adds 136 bits to it, bringing the sum total to 592.

However, each time slot on a TDMA frame is 0.577 ms long. This provides enough time for 156.25 bits to be transmitted (each bit takes 3.7 µs), but a burst only contains 148 bits. The rest of the space, 8.25 bit times, is empty and is called the Guard Period (GP). This time is used to enable the MS/BTS “ramp up” and “ramp down”. To ramp up means to get power from the battery/power supply for transmission. Ramping down is performed after each transmission to ensure that the MS is not transmitting during time slots allocated to other MS’s.

The output of burst formatting is a burst of 156.25 bits(one burst) or 625 bits(four bursts) for 20 ms sample. The transmission bit rate for GSM can be calculated to be 270.9 kbits/s(156.25/.577).

## STAGE 8: MODULATION & TRANSMISSION

The bits must then be sent over the air using a carrier frequency. As previously explained, GSM uses the GMSK modulation technique. The bits are modulated onto a carrier frequency and transmitted (e.g. 912.2 MHz).

The following figure summarizes the GSM transmission process.

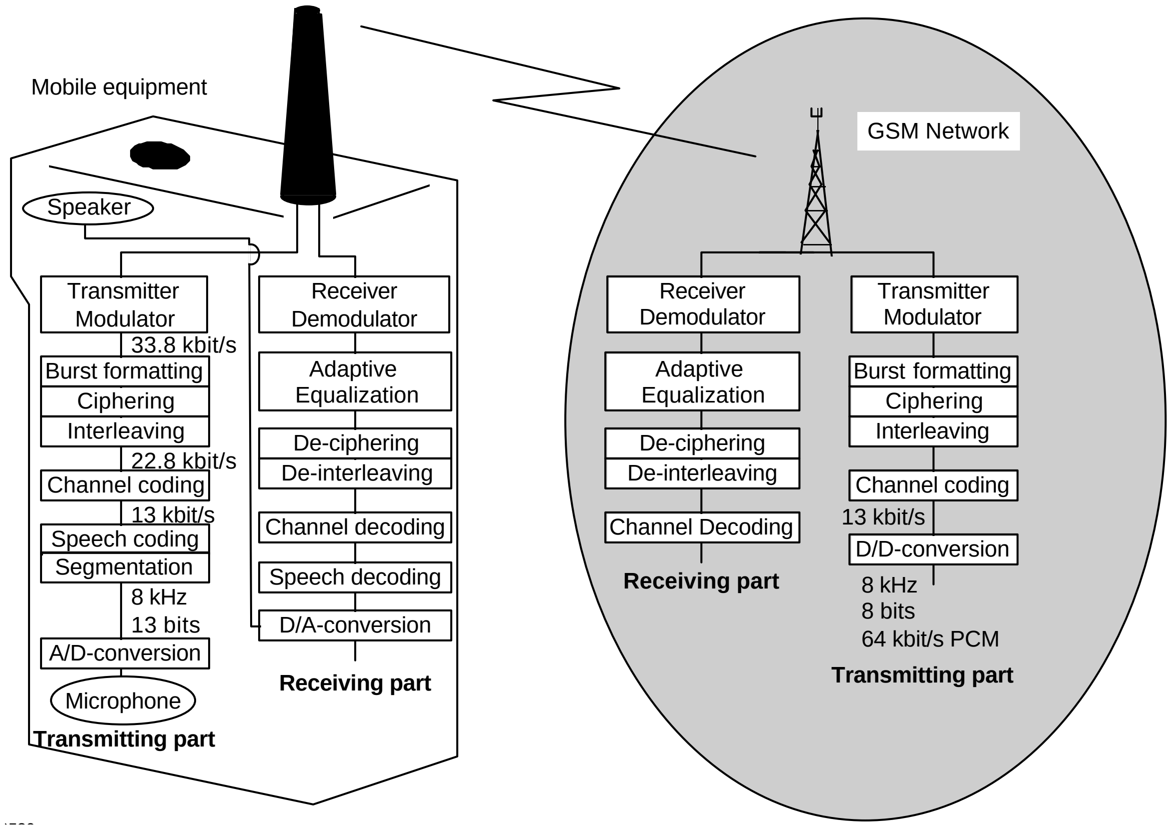
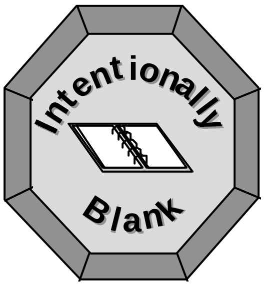



*Figure 3-33 GSM transmission process*


Intentionally
Blank