1

Introduction

1.1 Introduction to Telecommunications

Telecommunication systems in general consist of three main parts: transmitter; transmission media or channel; and receiver (see Figure 1.1). A more detailed description for each part now follows.

1.1.1 Transmitter

This unit is responsible for processing the data signal in order to be valid for the transmission media (channel). In its basic form the transmitter consists of three subunits: source data processing; data robustness; and signal modulation (see Figure 1.2).

In *source data processing* there are filters to determine the signal bandwidth; for an analog source and digital communication system there will be a sampler, analog to digital converters, and possibly data compression.

The *data robustness* subunit represents all processes performed on the data signal to increase its immunity against the unpredictable behavior and noise of the transmission media, therefore, increasing the possibility of successful reception at the receiver side. In wireless communication we transmit our data as electromagnetic waves in space which contain an almost infinite number of different signals. The ability of the receiver to receive our intended signal and clean it from associated interferences and noises, as well as distortion, is a real challenge. In digital communication systems this subunit may consist of channel coding and a pre-filtering process.

The aim of channel coding is to increase the reliability of the data by adding some redundancy to the transmitted information. This makes the receiver able to detect errors and even to correct them (with a certain capability). In digital communication systems we convert the source analog signal to a finite number of symbols. For example in the binary system there are only two symbols S_0 and S_1, or more popular 0 and 1. Generally we may have any finite number M of symbols, so that it is called an M-ary digital communication system. Those symbols are also known precisely at the receiver. The information flow is determined by the arrival sequences of these symbols. This is one major strong feature of digital systems over analog.

Systems Engineering in Wireless Communications Heikki Koivo and Mohammed Elmusrati

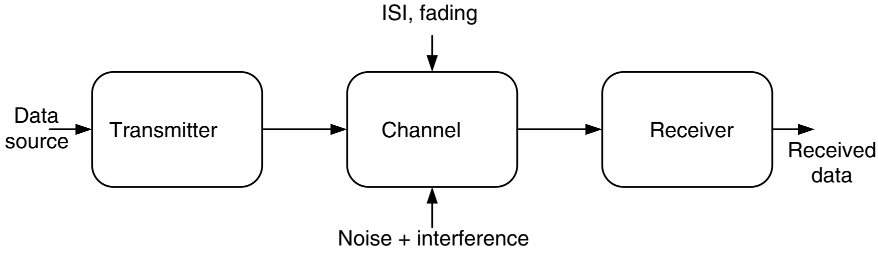

Figure 1.1 Simplified block diagram of communication system

Another possible way to increase the reliability of the transmitted signal is by making the symbols as unlike each other as possible. This increases the possibility of correct detection of the transmitted symbol. The pre-filtering process is a possible action done at the transmitter to minimize the effects of the channel distortion on the signal. Simply, if we can achieve pre-distortion of the signal, which has the reverse action of the channel distortion, then it is possible to receive the signal as distortion-free. Of course, this is only theoretical, but there are still several practical systems using this pre-filtering to reduce channel distortion effects.

Most of these systems require a feedback channel in order to adapt the characteristics of the pre-filter according to the channel behavior. One main type of signal distortion of the channel is due to the multipath reception of the transmitted signal.

The last part of this general presentation of the transmitter is the *modulation* subunit. We may express this subunit as the most important one for multiuser wireless communications systems. Usually the source data signal occupies a certain baseband bandwidth, for example for voice it could be between 20 Hz up to about 5 kHz. For the whole audible range the spectrum could be extended up to about 20 kHz. In wireless communications, it is not possible to connect this baseband signal directly to the transmission antenna. There are many reasons for this, such as lack of multi-usage of the channel, a very long antenna would be needed (antenna length is related to the signal wavelength), and non-suitable propagation behavior.

Modulation is the process of carrying information signals over a certain well-defined carrier signal. This carrier is usually a sinusoidal signal. Remember that the sinusoidal signal is the only signal which has zero bandwidth. In the spectrum domain, it is represented as a delta function located at the frequency value of the signal (and its mirror). Any sinusoidal signal is defined by three parameters: amplitude; frequency; and phase. It is possible to carry the data signal over any of those parameters, therefore we have three types of modulation techniques: amplitude modulation (or amplitude-shift keying in terms of digital communications

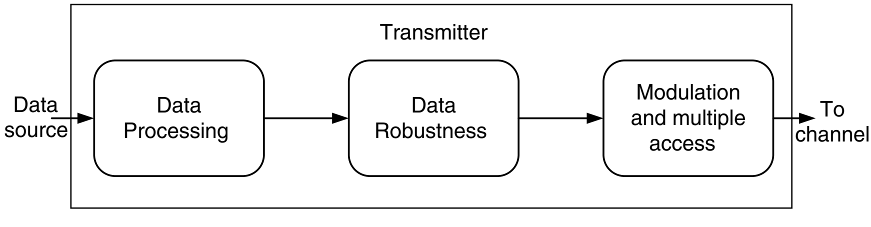

Figure 1.2 General representation of transmitters

terminologies); frequency modulation (or frequency-shift keying); and phase modulation (or phase-shift keying).

In digital wireless communications systems, we observe that phase-shift keying (PSK) is the most famous modulation technique. The reason is that PSK offers very high spectrum efficiency. For example, if the channel bandwidth is only 1 MHz, then after modulation (double sidebands) we can send a maximum rate of 1 Msymbol/s over this channel to be able to mitigate the intersymbol interference (ISI) problem. Using PSK, if the SINR is already very high at the receiver side, we can send large numbers of bits (i.e., more information) per symbol. For example, if we send 5 bits/symbol (i.e., 32-ary) then the transmitted data rate is 5 Mb/s over only 1 MHz bandwidth. However, as we increase the number of bits per symbol, we reduce the possibility of detecting the symbols correctly.

It is also possible to join different modulation methods together, for example amplitude and phase modulation. This is called quadrature amplitude modulation (QAM). These relations will be investigated more throughout the book.

Another important task of the modulation subunit is the multiple access possibility. Multiple access means the possible multi-usage of the channel between many agencies, operators, users, services, and so on. There are different kinds of multiple access methods such as: frequency division multiple access (FDMA); time division multiple access (TDMA); code division multiple access (CDMA); spatial division multiple access (SDMA); and carrier sense multiple access (CSMA). These techniques will be illustrated later.

1.1.2 Wireless Channels

A channel is the media where the signal will pass from transmitter to the receiver. Unfortunately, this transmission media is not friendly! Our signal will suffer from several kinds of problems during its propagation such as high power loss, where the received signal power is inversely proportional to the square of the distance in free space. In mobile communications, the received power may be inversely proportional with distance with power 4. For example, if the transmit average power is 1 watt, then after only 20 meters from the transmitter antenna, the average received power can be in order of $1/20^4 = 6.25\,\mu\text{W}$! By the way, this high-loss characteristic gives the advantage to reuse the same frequencies after some distance with negligible interference. This is one form of spatial division multiple access.

There is also the problem of shadowing where the received signal power may fall considerably because of buildings, inside cars, and so on. The large-scale power path loss is related to:

$$L \propto \frac{\aleph}{d^{-n}} \tag{1.1}$$

where $\aleph$ is the shadowing effects, which can be represented as a random variable with log-normal distribution, d is the distance between the transmitter and the receivers, and n is the path-loss exponent which is usually between 2 and 4. Throughout the book, we sometimes refer to channel gain rather than channel loss. Channel gain is just the inverse of channel loss, i.e., $G = 1/L$. For mobile communications, the path-loss exponent 4 gives a good fit with practical measurements.

The signal itself reaches the receiver antenna in the form of multipaths due to reflections, diffractions, and scattering. The reason for this multipath is due to natural obstacles

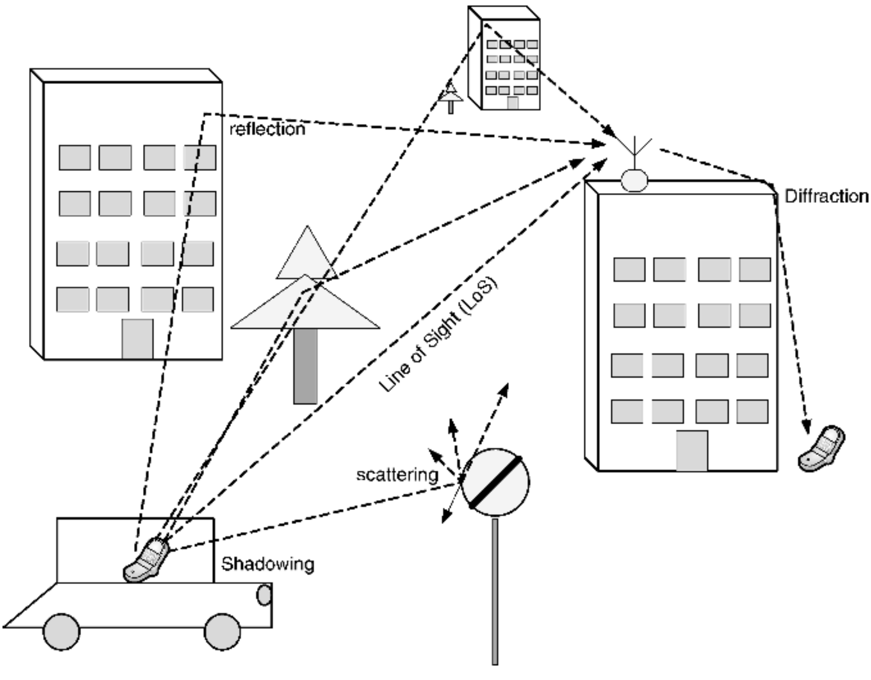

Figure 1.3 A simple picture illustrating the propagation conditions

(mountains, trees, buildings, etc.). Figure 1.3 shows a simple picture of the propagation conditions.

Most of the received signal comes through diffractions and reflections. It is rare that it follows a pure direct path, line of sight (LoS), except for point-to-point communication such as in microwave links. From Figure 1.3, assuming the transmitted signal in baseband to be $s(t)$, we may formulate the received signal as follows:

$$\begin{aligned} r(t) &= \mathrm{Re}\left(\left\{\sum_{k=1}^{B}\alpha_k(t)s(t-\tau_k(t))\right\}\mathrm{e}^{j2\pi f_c[t-\tau_k(t)]}\right)+n(t) \\ &= \mathrm{Re}\left(\left\{\sum_{k=1}^{B}\alpha_k(t)\mathrm{e}^{j2\pi f_c\tau_k(t)}s(t-\tau_k(t))\right\}\mathrm{e}^{j2\pi f_c t}\right)+n(t) \end{aligned} \tag{1.2}$$

where $\alpha_k(t)$ is related to the square-root of the path loss through path k and it is generally time varying, $\tau_k(t)$ is the delay of path k and it is also time varying in general, f_c is the carrier frequency, B is the number of paths, $n(t)$ is the additive noise and interference, and $s(t)$ is the transmitted signal in baseband. We will drop the additive noise part because we want to study the effects of channel propagations conditions.

Therefore, the equivalent received baseband is given by:

$$z(t) = \sum_{k=1}^{B}\alpha_k(t)\mathrm{e}^{j\varphi_k(t)}s(t-\tau_k(t)) \tag{1.3}$$

where φ_k is the phase offset. If the transmitter and/or receiver are moving with some velocity then there will be a frequency shift, which is known as the Doppler frequency, and is given by:

$$f_d = \frac{V}{\lambda} \tag{1.4}$$

where V is the relative velocity and λ is the wavelength. Including this Doppler frequency shift, the baseband of the received signal becomes:

$$z(t) = \sum_{k=1}^{B} \alpha_k(t) e^{j[2\pi f_d t + \varphi_k(t)]} s(t - \tau_k(t)) \tag{1.5}$$

Actually the Doppler effect is observed even for fixed transmitters and receivers. In this case the time-varying dynamic behavior of the channel is the source of this Doppler shift.

The effects of mobile channels on the received signal can thus be classified into two main categories: large-scale effects, where the received signal strength changes slowly (this includes distance loss and shadowing); and small-scale effects, where the received signal may change significantly within very small displacement (in order of the wavelength), and even without moving at all (because of the dynamic behavior of the channel). These dramatic changes in the signal strength may cause signal losses and outages when the signal-to-noise ratio becomes less than the minimum required. This is known as signal fading. There are two main reasons for small-scale signal fading. The first is due to the multipath and the second is due to the time-varying properties of the channel and is characterized by the Doppler frequency. The multipath characteristic of the channel causes fading problems at band-pass signal and ISI problems at the baseband. The following example explains in a simple way how this multipath may cause fading for the received signal.

Example 1.1

Study the effect of the multipath environment by assuming the transmission of an unmodulated carrier signal, $A\sin(2\pi f_0 t)$, $f_0 = 1.8$ GHz. Assume only two paths – one path causes a fixed delay and the other path a random delay. Plot the received signal power (in dB scale) using MATLAB® (a registered trademark of The Math-Works, Inc.) if the random delay has uniform distribution in the interval [0,1] μs.

Solution

Name the two paths' signals as $x_1(t)$ and $x_2(t)$, and the received signal $y(t) = x_1(t) + x_2(t)$. Also let the received amplitude be fixed (usually it is also random). Assume the amplitude of the signal received via the first path to be 1 V and via the second path 0.9 V. The signal received via the first path is $x_1(t) = A_1\cos(2\pi f_0[t - \tau_0]) = A_1\cos(2\pi f_0 t - 2\pi f_0 \tau_0) = A_1\cos(2\pi f_0 t - \theta_0)$, and from the second path is $x_2(t) = A_2\cos(2\pi f_0[t - \tau_2(t)]) = A_2\cos(2\pi f_0 t - 2\pi f_0 \tau_2(t))$. The total received signal is $y(t) = x_1(t) + x_2(t) = \cos(2\pi f_0 t - \theta_0) + 0.9\cos(2\pi f_0 t - 2\pi f_0 \tau_2(t))$, which can be simplified to (assuming $\theta_0 = 0$) $y(t) = \sqrt{1.81 + 1.8\cos(2\pi f_0 \tau_2(t))}\cos(2\pi f_0 t + \varphi_n)$. We can see clearly the random amplitude nature of the total received signal because of the random delay. The maximum amplitude is $\sqrt{1.81 + 1.8} = 1.9$ and the minimum is $\sqrt{1.81 - 1.8} = 0.1$. This indicates the high fluctuations of the received signal amplitude. The

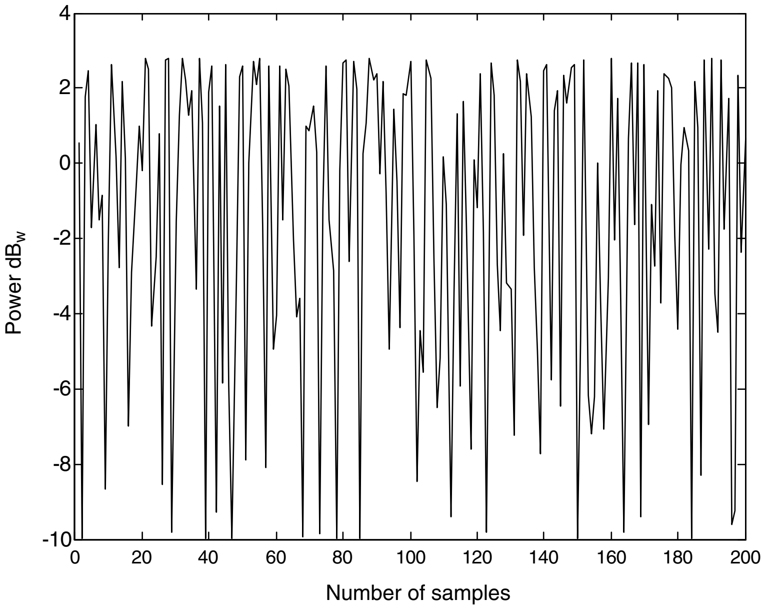

Figure 1.4 The effects of multipath on the received signal power

average received power can be expressed as $P_r(t) = [1.81 + 1.8\cos(2\pi f_0 \tau_2(t))]/2$. We may plot the received power in dBW as shown in Figure 1.4. The MATLAB® code is:

```
f0=1.8e9; % carrier frequency
tao=1e-6*rand(200,1); %generation of 200 random delay
Pr=(2+1.8*cos(2*pi*f0*tao))/2; % instantaneous power
Pr_db=10*log10(Pr); % expressing the power in dBw
plot(Pr_db) %plot the power
```

From Equation (1.5) we may deduce the impulse response of the wireless channel as follows:

$$h(t,\tau) = \sum_{k=1}^{B} \alpha_k(t) e^{j[2\pi f_d t + \varphi_k(t)]} \delta(t - \tau_k(t)) \tag{1.6}$$

If the symbol duration (T) is longer than the longest delay, i.e., $T > \max_k(\tau_k)$, we can approximate Equation (1.5) as

$$z(t) = s(t-\tau_0) \sum_{k=1}^{B} \alpha_k(t) e^{j[2\pi f_d t + \varphi_k(t)]} = \hat{\alpha}(t) s(t-\tau_0) \tag{1.7}$$

Therefore the received signal is simply the transmitted signal multiplied by the time-varying complex number representing the channel. Usually the relation between the received signal and the transmitted one is convolution because the channel is a linear time-varying system. Hence, the multiplicative relation shown in Equation (1.7) is a special case valid only when the signal duration is longer than the channel coherence time. This is known as the flat fading channel. The reason for this name is that when the channel loss is very high, i.e., $|\hat{\alpha}(t)|$ is very small, then the received symbol and all its replicas through the multipath will be faded at the same time.

The other type is called the frequency selective channel, which is obtained when signal duration is less than the channel delay profile. This leads to ISI between different symbols and causes large errors in the received signal. This problem sets an upper bound for the symbol rate that can be transmitted over wireless channels. However, it is interesting to note that this problem has been turned into an advantage by using CDMA.

With CDMA modulation, when the delay is larger than the duration of a single chip, the received replicas will have low correlation with each other. Using a Rake receiver, which consists of several correlators locked to each signal path, we obtain time diversity. In other words if one symbol is lost (because of deep fading), there is a chance of receiving the same symbol through another path.

Another way of mitigating the problem of ISI in frequency selective channels is by dividing the signal into many orthogonal parallel subbands and sending them simultaneously in the channel. Every subband will see the flat fading channel, and then at the receiver side we can recombine all subbands to their original band. A very efficient method to perform this process is called orthogonal frequency division multiplexing (OFDM).

The effects of time variations of the channel appear as a frequency deviation from the carrier frequency. For example if the transmitted signal is $\mathrm{Re}\{s(t)\mathrm{e}^{j2\pi f_0 t}\}$, it will be received as $\mathrm{Re}\{s(t)\mathrm{e}^{j2\pi(f_0 \pm f_d)t}\} = \mathrm{Re}\{s(t)e^{\pm j2\pi f_d t}\mathrm{e}^{j2\pi f_0 t}\}$. We have a time-varying phase which will be difficult to track if f_d has large value. The effect of the time-varying behavior of channels is known as slow fading (if f_d is small) and fast fading when f_d is large. More analysis about channels is given in Chapters 3 and 4.

We have so far discussed the effects of the inherent properties of wireless channels on signals. Moreover, there is an almost infinitely number of signals in the wireless channel. Some of those are manmade and other are natural signals from the sun, stars, earth, and so on. Usually we consider only signals which fall within or close to our signal bandwidth. Other signals can be easily removed using front-end band-pass filters at the receiver side. Signals from other communication systems within our bandwidth are known as interferences. Other signals are known as noise. Actually, one major noise part is generated within the receiver circuits. It is known as thermal and shot noise. The noise generated by the receiver is characterized by the noise figure or the equivalent noise temperature of the receiver. Usually, the noise figure of the receiver is one of the parameters that determines receiver sensitivity.

1.1.3 Receiver

The receiver is an electronic circuit which is used to recover the original signal from the intentional modifications (modulations, coding, ADC, etc.) and the channel inherent distortions (channels, noise, interferences, etc.). The receiver can be described as reversing or undoing the operations of the transmitter. In this sense the receiver can be simplified into three blocks as shown in Figure 1.5.

On the receiver side, we have band-pass filters to reject all signals other than the interested band. In multiuser communications, the receiver should receive only its intended signal through the multiple access method. Next we have the demodulation process to move the signal from band-pass to its original baseband form. Data recovery is the block which uses all data processing made at the transmitter to enhance reception reliability. For example, if the transmitter uses

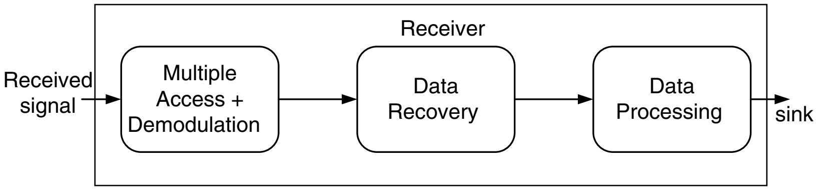

Figure 1.5 Simplified structure of wireless receiver

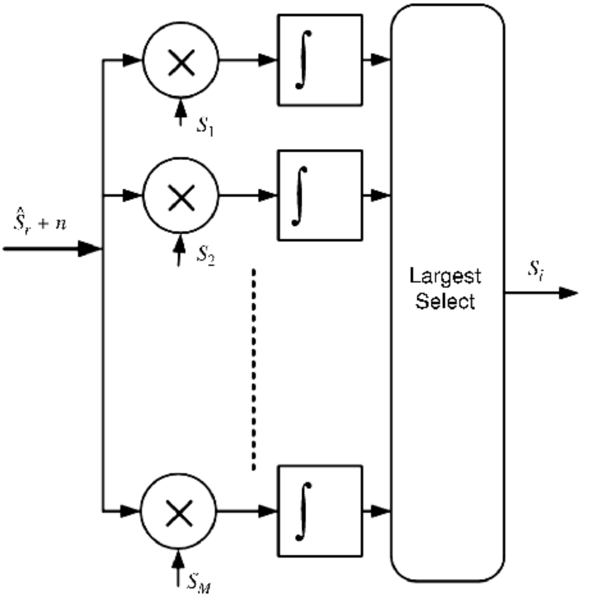

Figure 1.6 Correlator detection

forward error correction (FEC) coding, then in the data recovery block the decoding process will detect any errors and correct them if possible. Finally, in the data processing part we should retrieve the signal as close to its original form as possible. As mentioned before, in digital communications we send one of the known symbols at a time. The symbols are known precisely at the receiver as well. The information flow is in the transmission sequence of those symbols. From this, the optimum detector at the receiver is based on a correlation process. The received symbol will be distorted according to the channel and also corrupted by noise and interference. The receiver will measure the matching between the received symbols with all possible symbols using correlators and select the one which has the highest match score as shown in Figure 1.6. Finally, the detected signal is delivered to the sink (speaker, screen, computer input port, etc.).

1.2 The Quality of Service (QoS) Parameters

In wireless communications, it is important to define a set of parameters to measure the communication quality. The QoS parameters can be classified into two main parts. One part can be sensed by the end user such as voice quality, blocking and outage probability, and data rate. The second part is related to the security and privacy which usually cannot be sensed by the end user during the call. Operators should offer accepted quality of service and at the same time keep the security and the privacy of their customers. Feedback control can be applied to improve QoS. To assess the link quality, measures of performance are needed. Some of the QoS parameters can serve as such. One of the most important QoS parameters is the ratio of the average power of the desired signal - to - the average power of the interference and noise signals. It is commonly known as the signal to interference and noise ratio (SINR). Other names are possible such as signal-to-noise ratio (SNR), where noise is the dominant part; signal-to-interference ratio (SIR); where the interference is dominant; and carrier-to-interference ratio (CIR), which is used to indicate the ratio at band-pass signal. These different names will be used in this book interchangeably. There is one problem with the SINR indicator when it is applied to indicate the quality of digital modulated signals. The SINR does not consider the number of bits per symbol in multilevel modulation, which may lead to wrong conclusions. This is explained in the next example.

Example 1.2

Assume PSK modulation where the received signal is $x(t) = 3\cos(2\pi f_0 t + \phi_i(t)) + n(t)$ and the average noise power is 0.1 W. The input resistance of the receiver is 1 Ω. Compute the SNR for binary PSK (BPSK), where we have only two symbols and 32-PSK, where every symbol carries 5 bits.

Solution

We can calculate the SNR simply as the ratio of the average received power to the average noise power corrupting the signal such as $SNR = \frac{9/2}{0.1} = 45 \Rightarrow SNR_{dB} = 10\log_{10}(45) = 16.5\ dB$. This SNR value is the same for both cases of modulations (binary and 32-ary)! Since we do not increase the transmission power, it is clear that the probability of error will be greater when we send more data per symbol. This can be verified by studying the constellation map where all the symbols are positioned. The distance between the symbols becomes shorter as we increase the number of symbols without increasing the transmission power. The possibility of incorrect detection now becomes much greater because of noise. This fact can be observed in the bit error rate formulas of different modulation types.

We can make the SINR indicator more relative to digital communication systems by computing it at the bit level. In other words we compute the average received power per bit which can be done by multiplying the average received power with the bit duration. From physics we know that the product of power with time is energy. Therefore we can compute the received bit energy E_b as $E_b = ST_b$, where S is the average received signal power and T_b is time duration per bit. This can be made more understandable by forming a unitless expression dividing E_b with the noise energy, known as noise spectral density N_0, $N_0 = \delta_n^2/BW$, where δ_n^2 is average noise power and BW the bandwidth. The unitless expression is then E_b/N_0, which is useful when comparing different communication techniques having different modulation levels. It is related to SINR by the following expression $\frac{E_b}{N_0} = SINR\frac{BW}{R_b}$, where R_b is the bit rate and BW is the bandwidth, which can be the same as the symbol rate for some systems. Does the end user notice the value of the SINR? Actually, not directly! The end user notices more the block error rate (BLER). When BLER is large then there will be outage, voice degradation, low throughput (because of retransmissions), and so on. However, there is a direct relation between BLER and SINR. In AWGN channels the BLER decreases when the SINR increases. This mapping can be given in closed-form mathematical expression in some simple cases such as when noise is additive white noise or when the fading is modeled mathematically. However, in reality, these give only rough estimates of the real situation.

Exercise 1.1 What is the difference between bit error rate (BER) and block error rate (BLER), and what are the effects of coding in this case? *Hint*: see Chapter 8.

The SINR is used directly as an indicator for the channel quality. Since the channel condition is measured generally by the BER or BLER after coding, why do we not use them directly as channel indicators instead of SINR or E_b/N_0? The answer is that the required estimation time is too long. The SINR estimation can be done in a very short time, but a direct measurement of the BLER or BER may require a longer time, which leads to slow adaptation rate of the resources. For example, assume that the target BER is 10^{-3}, and the data rate is 10 kb/s, so that we will need to test about 10 000 bits to have robust estimation for the BER. This number of bits will require 1 s of monitoring time. This is a very long time in terms of resources adaptation. For example, in UMTS mobile systems, the power adaptation rate is 1500 Hz, i.e., the power is adapted once per 0.667 ms. Therefore we need a very fast channel quality estimator. This can be obtained with SINR (or E_b/N_0) measurements.

The throughput is another important parameter of the QoS in wireless communication systems. Throughput is an indicator for the actual information rate and it is different than the data rate. The data rate is the actual transferred data per unit time. This data contains (besides the required actual information) other types of bits such as coding bits, destination address header, and signaling protocol bits. Moreover, packets could be retransmitted if they were not correctly decoded. We can say that throughput is usually less than the data rate. Although the other bits are necessary to perform the communication between the transmitter and the receiver, the end user does not really care about them. We are much more interested in the actual information rate, for example how fast we can download certain files wirelessly through our mobiles.

Packet delay is another QoS parameter which each operator should specify accurately. When we look at the delay between data transmission and reception (end to end) we may call such delay the latency. In real-time applications maximum latency should be strictly specified, at least within a certain confidence interval, for example, 99% of the transmitted packets will have delay less than 30 ms. There are several reasons for packet delays in wireless communications such as protocol delay (e.g., in CSMA/CA), queuing delay (because of limited resources), retransmission delay (according to error reception), and propagation delay (significant in satellite links). When the delay exceeds some limit we will have packet loss which should be also specified in the QoS agreement. There is another problem with delays when they are not fixed, i.e., when the delay period is a random variable. This introduces something known as jitter, which is computed as the maximum delay minus the minimum delay. When the jitter is longer than the packet period, there is a high chance that the packets will not arrive in the correct order as well as interfere with others. Moreover, in certain applications such as wireless automation, it would be difficult to design high performance and stable control systems in the presence of a wide range of random delays.

Based on the above summary of the QoS parameters, we may state that the required QoS depends on the application. There are several classifications of applications such as real time and non-real time, fixed and elastic and so on. Let us take few simple examples for the required QoS parameters with different applications.

- **Voice communication** (GSM): fixed and low throughput ($\sim$13 kb/s), strict maximum delay, relaxed BER ($\sim 10^{-3}$), and relaxed packet loss probability (our brains are excellent at data fitting, so that the lost packets may not be missed if they are not too large). This is considered as a real-time application.
- **SMS messages** (non-real-time application): very low throughput, very relaxed delays, strict maximum allowed BER (we do not like to receive a different message than has been sent!).
- **Internet exploring** (large file download): the higher the throughput the better, relaxed delays, and strict BER. Here we are more interested in high throughput in the average sense. For example if a file can be downloaded within 30 s, it is much better than 3 min regardless of the continuity of the service. Let's explain this more. Assume that during this 30 s you get served only 1 s every 9 s (i.e., three times) with a data rate of 10 Mb/s. This means that during these three times you will download 30 Mb. On the other hand, if you have fixed continuous services but with a data rate of 30 kb/s, then you will need a total time of 1000 s to get this 30 Mb! Hence, in these types of applications, we do not care about the continuity of the service as long as we have a high throughput on average.
- **High secure applications** if we want to make a bank transfer, then we may relax the throughput as well as the delay, but we should have very low BER and the highest security and privacy.

From the above we see that the QoS requirements depend mainly on the applications.

1.3 Multiple Access Techniques

Multiple access methods are critically required for the multi-usage of the channel. In other words, how can several transmitter/receiver pairs use the same available channel with minimum interference between them. Consider a real situation which describes the multiple access need and methods.

Assume a classroom with many students (from different countries) and one teacher. When the teacher is talking and all students are listening, we have a broadcasting situation where we do not need to consider multiple access. Assume one student cannot hear clearly, then she may ask the teacher to increase his voice level. This can be considered as a feedback channel to improve the SNR at the receiver side. Suppose the teacher divides class into random pairs (not neighboring pairs), where each pair should talk to each other. What do you think will happen in this case? Every pair will start talking, but because of the interference from other pairs, each pair will try to talk more loudly (increasing the transmission power) and use the available degree of freedom to increase the possibility of correct reception such as beamforming (through adjusting the location of ears in the right direction) and image recognition (through looking at the mouth of the intended speaker). When the number of pairs is large within the limited size of the classroom, you can imagine that it will be chaos. What are the methods to make the multi-usage of the limited resources (bandwidth) more organized and successful?

For the first method, the teacher can divide the time between all pairs. Hence, during the first time slot the first pair talks freely without interference, and then they become silent during the next time slot, where the channel is given to the next pair and so on. In telecommunications this is called time division multiple access (TDMA). Can you suggest another way? Suppose every pair speaks a different language (e.g., Arabic, English, Finnish, French, etc.), which cannot be understood by other pairs. Therefore all pairs can talk simultaneously at just the proper voice level to minimize the interference to others. We can filter out the non-understandable languages. This is also used in telecommunications and known as code division multiple access (CDMA), where every terminal uses a special code which is almost orthogonal to other codes.

Another method is if we can reorganize the pair locations in the classroom so that every transmitter/receiver pair becomes very close to each other and each speaks with a very low voice, then all pairs can communicate simultaneously. In telecommunications, we call this spatial division multiple access (SDMA), where we reuse the same bandwidth keeping a proper distance between pairs.

And another way? Yes, our normal way of talking. Simply, the speaker who wants to say something to his partner, will listen to the channel. If no one is talking, then she can start talking. As soon as she starts talking, no one else will talk or interrupt until she completes the message (a polite environment!). Then it becomes free for others to talk in the same manner. If, at that moment two persons start talking simultaneously, then both will stop and be quiet for a while. One of them may start after some random time. This method is unorganized multiple access and it is widely used in ad hoc networks. In telecommunications it is known as carrier sense multiple access with collision avoidance (CSMA/CA).

The problem of CSMA/CA is that in general no target QoS can be guaranteed, however, there are more advanced forms which may guarantee a minimum QoS (based on user demand and channel situation) such as the multiple access methods proposed for WiMAX.

Are there other ways? From our side it is enough to show these examples of analogies between multiple access techniques in telecommunications and human communication. There are several other methods for multiple access methods. The frequency division multiple access (FDMA) may be the first technical method applied in telecommunications. In FDMA the available bandwidth is divided between users simultaneously. However, this method is not widely used in modern communication systems because of its lack of efficiency where we need to keep a guard band between different signals to minimize the interference. Its efficient version, OFDM, is much more popular and can be found in many recent standards such as WiFi, WiMAX, and cognitive radios. Furthermore for the spatial division multiple access, there are efficient methods using beamforming principles which can greatly enhance the system capacity as will be described in Chapter 9.