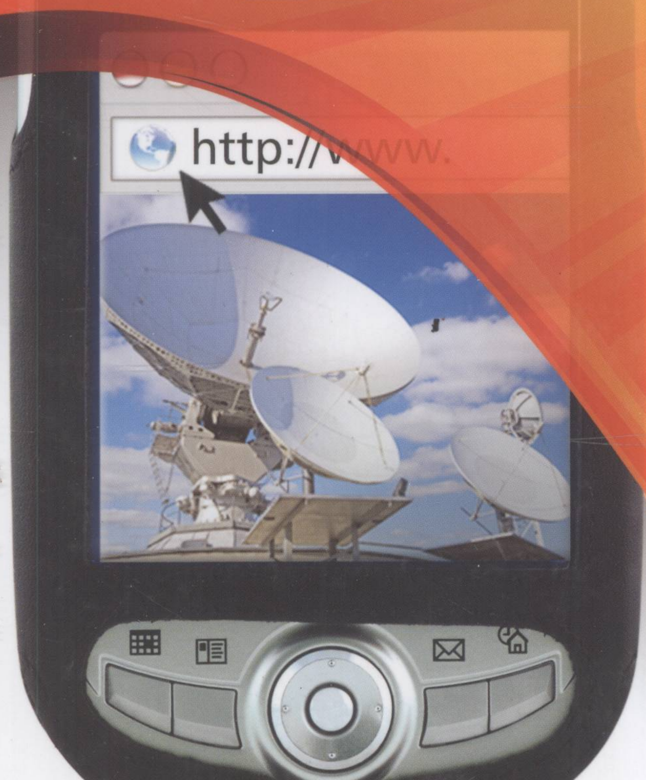
Systems Engineering in Wireless Communications



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SYSTEMS ENGINEERING IN WIRELESS COMMUNICATIONS

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Preface

In Helsinki during a visiting lecture, an internationally well-known professor in communications said, 'In the communications society we have managed to convert our proposals and ideas to real products, not like in the control engineering society. They have very nice papers and strong mathematics but most of the real systems still use the old PID controllers!'. As our background is mainly in control as well as communications engineering, we know that this thought is not very accurate. We agree that most of the practical controllers are analog and digital PID controllers, simply because they are very reliable and able to achieve the required control goals successfully. Most of the controllers can be explained in terms of PID. The reasons behind this impressive performance of PID will be explained in Chapter 2.

There is a hidden fact that many researchers and professionals do not pay enough attention to. This is that control engineering is one of the main pillars of modern communication systems in both hardware and algorithms. On the hardware front, we know the importance of the phase-locked loop (PLL) and its modified version, the delay-locked loop (DLL), in the implementation of communication receivers. We can say that phase-locked loops are an essential part of all kinds of today's communication receivers. These systems are simply small closed-loop control systems. From an algorithmic point of view, the multiple access control (MAC) layer is based on control theory concepts. Some examples are power control, rate control, scheduling, handover, and generally radio resource management algorithms. When there is feedback measurement and action based on that measurement, then there is control engineering. We may generalize this view to include all kinds of adaptive systems such as equalizers, beamforming antennas, adaptive modulation and coding. Therefore, if we think of the modern communications system in terms of the human body, then control algorithms are the heart and communication algorithms are the blood. This is a general view of the contribution of control engineering in telecommunications. Moreover, if we extend this vision to include system engineering, which means in addition to control engineering, system identification and modeling, the contribution is much more. Channel identification and modeling is one major requirement in the design of transmitters and receivers. However, this inherent integration between communication and control is not that visible for many people. This is one reason why you have this book in front of you now. We have tried to show some of the applications of control engineering in communication systems, but we did not want to just collect algorithms from literature and bind them together into a reference book. We wanted to generate one easy-to-read book which is accessible to a wide range of readers from senior undergraduate students up to researchers and site engineers. To achieve this target, we have included many numerical examples and exercises. A key approach to becoming more familiar and understanding the deep secrets of sophisticated control algorithms is through simulation. Therefore MATLAB®/Simulink® is used as a tool to illustrate the concepts and shows promise in a diversity of ways to attack problems. We believe that Simulink® can provide a more transparent way of studying, for example in the simulation of self-tuning estimators and controls and Kalman filters.

We have many people to thank for this book.

Heikki Koivo wants to thank his students Jani Kaartinen, Juha Orivuori and Joonas Varso, who helped in setting up the numerous MATLAB®/Simulink® examples. He is eternally grateful to his wife, Sirpa, who suffered the pain caused by writing and the time spent apart. Special thanks go to Jenny and Antti Wihuri Foundation and the Nokia Foundation who have supported the author with stipends.

Mohammed Elmusrati wants to thank his friend Dr. Naser Tarhuni for his constructive comments about the book. He would also like to thank his wife Nagat and daughters Aia and Zakiya for their care, patience and support. Special thanks to his father Salem Elmusrati for his continuous moral support and encouragement.

The Co-author Mohammed Elmusrati wishes to make it clear that section 8.2 of this book is reproduced from his Ph. D. thesis entitled: *Radio Resource Scheduling and Smart Antennas in CDMA Cellular Communication Systems*, ISBN 951-22-7219-9 (published by Helsinki University of Technology - Finland), and not from a work entitled *Advanced Wireless Networks: 4G Technology* written by Savo Glisic which in section 12.5 reproduces part of Elmusrati Ph.D. Thesis.

As stated in this book, proper feedback is critical for system performance and efficiency. Hence, your feedback is very important to us. If you have any comments, corrections or proposals, please do not hesitate to contact us on the following email address: sys.eng.wireless.comm@gmail.com.

List of Abbreviations

AWGN	Additive White Gaussian Noise
AM	Amplitude Modulation
ASK	Amplitude Shift Keying
ARMA	Auto Regressive Moving Average
ARMAX	Auto Regressive Moving Average Exogenous
ARX	Auto Regressive Exogenous
BER	Bit Error Rate
BLER	Block Error Rate
BIBO	Bounded Input Bounded Output
BPF	Band-Pass Filter
BPSK	Binary Phase Shift Keying
CAC	Connection Admission Control
CDMA	Code Division Multiple Access
CIR	Carrier to Interference Ratio
CMTTP	Centralized Minimum Total Transmitted PowerCSMA
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CSOPC	Constrained Second-Order Power Control
DBA	Distributed Balancing Algorithm
DCPC	Distributed Constrained Power Control
DDMA	Demand Division Multiple Access
DL	Downlink
DoA	Direction of Arrival
DPC	Distributed Power Control
DPSS	Discrete Prolate Spheroidal Sequence
ESPC	Estimated Step Power Control
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
FER	Frame Error Rate
FFT	Fast Fourier Transform
FLPC	Fuzzy Logic Power Control
FMA	Foschini and Miljanic Algorithm
FSPC	Fixed-Step Power Control
GMVDR	General Minimum Variance Distortionless Response
GUI	Graphical User Interface

HSDPA	High Speed Downlink Packet Access
HSPA	High Speed Packet Access
IFFT	Fast Fourier Transform
ISE	Integral Square Error
ISM	Industrial, Scientific and Medical
ITAE	Integral Time Absolute Error
LoS	Line of Sight
LS-DRMTA	Least Square Despread Respread Multitarget Array
LTV	Linear Time Variant
MIMO	Multiple-Input Multiple-Output
MISO	Multiple-Input Single-Input
MLPN	Multilayer Layer Perceptron Network
MMSE	Minimum Mean Square Error
MO	Multiple Objective
MODPC	Multiple Objective Distributed Power Control
MTPC	Maximum Throughput Power Control
MVDR	Minimum Variance Distortionless Response
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
OLPC	Outer-Loop Power Control
PAN	Personal Area Network
PD	Proportional Derivative
PI	Proportional Integral
PID	Proportional Integral Derivative
PLL	Phase-Locked Loop
PSK	Phase Shift Keying
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
RBFN	Radial Base Function Network
RLS	Recursive Least Squares
RRM	Radio Resource Management
RRS	Radio Resource Scheduling
RX	Receiver
SDMA	Spatial Division Multiple Access
SDMPC	Statistical Distributed Multi-rate Power Control
SDR	Software-Defined Radio
SIMO	Single-Input Multiple-Output
SINR	Signal to Interference and Noise Ratio
SIR	Signal to Interference Ratio
SISO	Single-Input Single-Output
SMIRA	Stepwise Maximum Interference Removal Algorithm
SMS	Short Message Service
SNR	Signal to Noise Ratio
SOR	Successive Over-Relaxation
SPC	Selective Power Control
SRA	Stepwise Removal Algorithm

STPC	Self-Tuning Power Control
STPPC	Self-Tuning Predictive Power Control
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TX	Transmitter
UMTS	Universal Mobile Telecommunications System
VCO	Voice Controlled Oscillator
WCDMA	Wideband CDMA

Contents

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.

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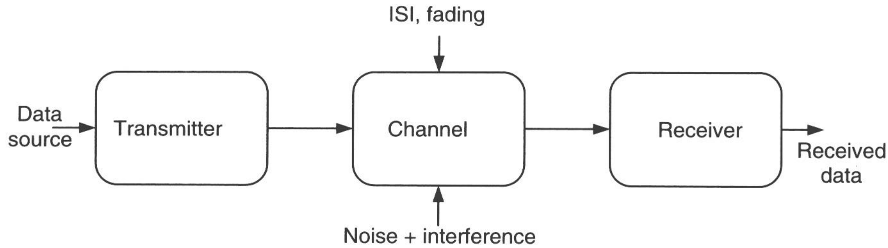

Figure 1.1 Simplified block diagram of communication system

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

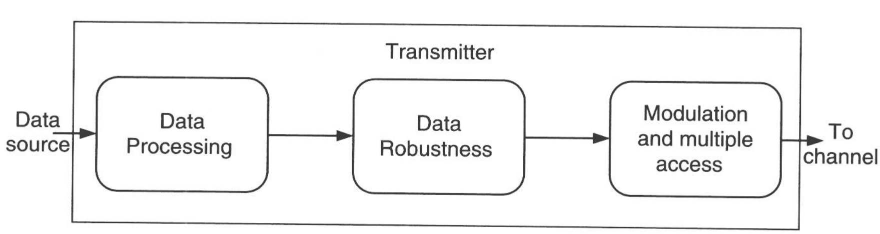

Figure 1.2 General representation of transmitters

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

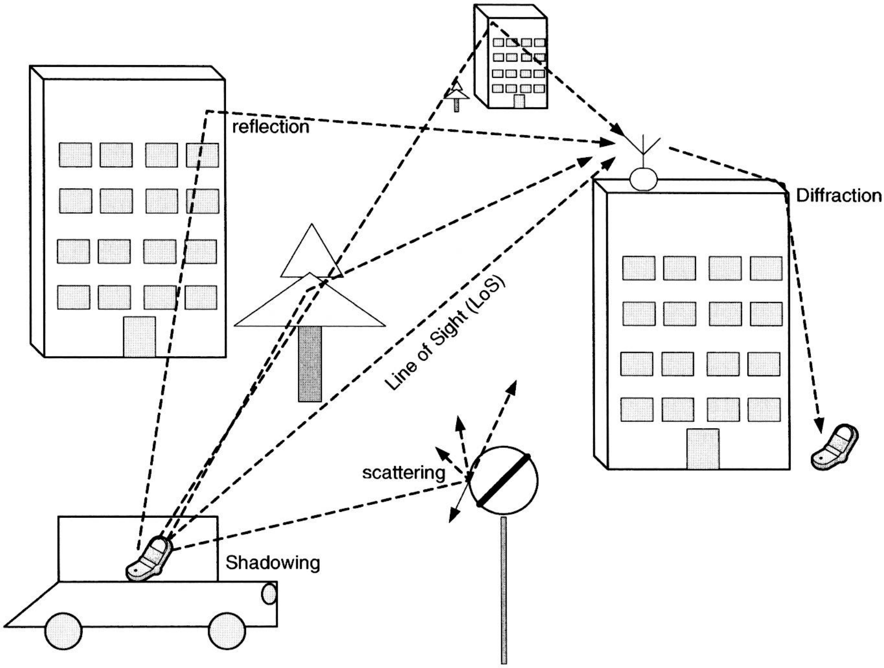

Figure 1.3 A simple picture illustrating the propagation conditions

(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