Fundamentals of Information and Communication Technologies

Fundamentals of Information and Communication Technologies

By

Shun-Ping Chen

Cambridge Scholars Publishing

Fundamentals of Information and Communication Technologies

By Shun-Ping Chen

This book first published 2020

Cambridge Scholars Publishing

Lady Stephenson Library, Newcastle upon Tyne, NE6 2PA, UK

British Library Cataloguing in Publication Data
A catalogue record for this book is available from the British Library

ISBN (10): 1-5275-5585-2
ISBN (13): 978-1-5275-5585-3

CONTENTS

PREFACE

Information and Communication Technologies (ICT) have developed rapidly in the last decades, with performance improved significantly by using sophisticated software and hardware techniques, evolving from analogue technologies to efficient, reliable and inexpensive digital technologies.

The goals of the information and communication technologies are to efficiently process the multimedia and other data information and to reliably transmit them from the information source to the destination, or technically speaking, from the transmitter to the receiver. This overall end-to-end process involves many disciplines, like multimedia source coding, channel coding, modulation and wireline and wireless transmission. Each of these disciplines have been treated in numerous textbooks and taught in numerous courses at universities.

The author has been teaching various Bachelor and Masters courses at the Darmstadt University of Applied Sciences (Hochschule Darmstadt) in Germany since 2008. He is very lucky to be involved in many lectures directly related to the above-mentioned information and communication technology fields, leading to the idea to write a textbook, which does not primarily go into too much detail on all of these exciting fields, but rather provides an overview of the end-to-end information processing and communication.

This textbook is designed not only as lecture notes for students of Bachelor and Masters programmes at universities, but also for interested researchers and engineers, who would like to get an overview and cover the most fundamental aspects of information and communication technologies and networks, as well as for all other interested people in the scientific community and society. For this purpose the author also tries to describe the technical aspects more comprehensively on one hand, and on the other hand also focuses on his teaching and research areas like antenna arrays and fibre and free space optical communications as well as network security.

The author appreciates many fruitful discussions with colleagues, students and research partners. Especially, he would like to thank his colleague, friend and advisor Professor Heinz Schmiedel for proofreading the manuscript, for many helpful advices and for his continuous support and encouraging discussions.

The author would like to thank Laurence Fenton for careful proofreading and Rebecca Gladders for many helpful advices and support.

CHAPTER 1

INTRODUCTION

Following on from the development of wireline and wireless technologies in the nineteenth century, such as the telegraph and the telephone, the modern information and communication technologies witnessed tremendous advances in the twentieth century, from the inventions of transistors, computers, laser diodes and satellite communications to digital cellular mobile networks, optical fibres and, last but not least, the internet. All are being permanently further developed and continuously improved.

Starting with the basic architecture of the information and communication technologies and networks, a goal is a reliable, high-quality communication link for transmitting the multimedia information data between the source of the information and the destination. To ensure high availability and reliability we want low bit error rates, low blocking probability, low delay and low jitter in the ideal case. The applications could be conversational phone calls, video conferences, internet browsing, video streaming, IP television, multimedia data transfer, etc. In the upcoming 5G mobile communications, new promising applications like eMBB (enhanced Multimedia Broadband), mMTC (massive Machine Type Communications, like sensor applications) and uRLLC (ultra-Reliable Low-Latency Communications) will be enabled.

In daily life, different types of information, like classical text, still images, voice, data, music or motion pictures and video streaming, will be processed and transmitted. This kind of communication is defined as H2H (human-to-human) communication.

Along with the rapid improvement of digitalisation and new applications in industrial branches like autonomous driving, power grids, smart factories, smart homes, intelligent traffic management and the internet of things (IoT), where an extremely large number of sensors will be applied, the measurement and control data must be transmitted reliably and with extremely low delay, especially for so-called mission critical applications. Therefore more and more machine type communications (MTC) will be indispensable. Different types of sensors can detect parameters like temperature, forces, velocity, acceleration, humidity, etc.

Also bio-chemical sensors for detecting the content of certain chemical components, blood glucose, taste, odour, flavour, etc., are now being developed for the healthcare, medicine and chemical industries. These machine type communication technologies provide the possibility for automatic data processing in relevant fields like autonomous driving, health care, traffic management, smart homes, smart grids, smart factories and so on.

The challenging tasks for the information and communication engineers will be to design intelligent systems to convert these primary physical and chemical parameters to electrical information data sources, usually in digital, binary format, which can then be transmitted from the information source to the central information processing unit, and then further to the receivers by utilising sophisticated digital communication networks and technologies.

The transmission can take place via wireline systems (copper wires, optical fibres, coaxial CATV infrastructure and hybrid fibre coaxial HFC as well as the 50/60 Hz power lines that are partially used to transmit digital information in the higher frequency bands) or wireless communication systems (WLAN/WiFi, Bluetooth, ZigBee, RFID, cellular mobile networks like GSM, UMTS, LTE, satellite and future 5G mobile networks).

Driven by the invention of transistors, laser diodes, optical fibres, cellular mobile network concepts and the internet in the twentieth century, traditional analogue communication technologies have been almost completely replaced by digital communication technologies. Therefore in this book we only focus on digital information and communication technologies.

Chapter 2 discusses the fundamentals of information theory.

In Figure 1 the end-to-end digital communication channel model from the information source to the destination, or (respectively) from the transmitter to the receiver, is illustrated schematically. Prior to the source coding, the multimedia information sources for H2H and MTC will be first converted from the analogue information sources (voice, data, images and moving pictures, etc.) to a digital information data stream in terms of binary bits by using an ADC (Analogue Digital Converter).

These digital, binary data streams will be optimised by compressing techniques, where the redundancy and irrelevance will be removed, in order to efficiently utilise the scarce network bandwidth in the transmission networks. This process is called source coding, and will be discussed in Chapter 3.

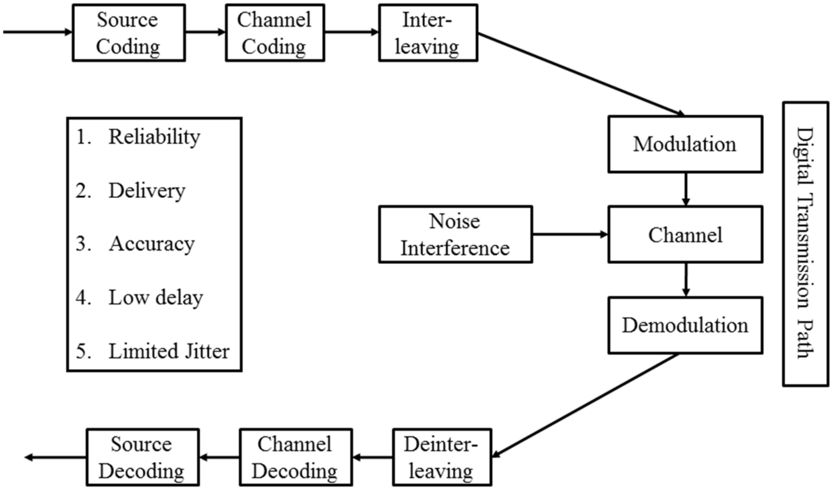


Figure 1 Digital communication channel model

The source coding techniques of the multimedia information sources will be discussed, in order to understand how they achieve the best performance with the best reliability for the limited physical transmission network resources. If necessary, the source-coded information will be further encrypted in order to guarantee confidentiality. The integrity of the data will be checked by using the hash value, which is like the fingerprint of the information content.

In the second step some useful redundant bits and bytes are added, in order to enable FEC (forward error correction). This can be done by sophisticated channel coding schemes like Viterbi coding and convolutional coding. After that, interleaving is used to separate the neighbouring bits and reorder them in a proper manner, in order to separate the burst errors to single bit errors, and thus to make the FEC more efficient.

Source coding reduces the data volume to a minimum, whereas channel coding is used to enable automatic bit error correction, therefore to guarantee the performance and reliability of the transmission networks.

This digital information format will be modulated from the base band to the corresponding carrier frequency for certain physical media like wireline, or free space by using properly defined modulation schemes to achieve high bit rates, with low bit errors and low delay. The frequency could be f = 900 MHz, 2 GHz for the classical cellular mobile communications in the form of electromagnetic wave propagating in the free space, and approximately f = 193.5 THz (λ = 1.55 μm) for the guided wave propagation in optical fibres. During the transmission either in free space or in the wirelines like optical fibres, different mechanisms lead to the gradual reduction of the signal power level from the transmitter to the receiver (attenuation), the spreading of the binary signal pulses in the time domain (dispersion) and the disturbance of the information signal by interferences by other users or other disturbing noises in the same channel or the same frequency band, so that the distance between the transmitter and receiver, and achievable bit rates, will be always limited by the signal power level relative to the noise or interference. At a certain threshold, also called receiver sensitivity, the signal as binary 0s and 1s can hardly be recognised and received anymore without bit errors. Therefore so-called repeaters and regenerators to refresh, regenerate and re-amplify the signal are used to increase the difference between the wanted signal power level and the noise or interference power level. The ratio between them is also defined as SINR (signal to interference and noise ratio). The better the SINR, the better the channel quality will be, and correspondingly a higher throughput will be achieved.

The quality of such a transmission channel is generally characterised by the fundamental technical parameters like reliability, availability, bit error rates, accuracy, delay or jitter (variation of the delay), etc. Besides other network management measures, it is important to reduce the time needed for repairs in case of failures, outages and performance degradation of single network elements, network sections or application servers.

At the receiver site, the information signal will be transformed back to the format of the source step by step. Firstly the binary information bits and bytes are demodulated back to the base band, de-interleaved back to the original order, channel-decoded by performing the FEC and then source-decoded. The last step is to convert the digital data stream to the analogue information. The digital binary bits and bytes will then be converted to the analogue information – i.e. text, language, music, video, measurement data, etc. – using a DAC (Digital Analogue Converter).

Concerning the transmission path, the medium could be free space or air, by utilising the free space propagation of electromagnetic waves (like GSM, UMTS, LTE, Wireless LAN/WiFi, satellite communications, radio, TV and free space optical communications (FSOC)), or by guiding the electromagnetic waves in metallic waveguides or optical fibres consisting of silica quartz glass. For lower frequencies, the classical twisted copper wires, either schielded or unschielded, can be used, for example, in the xDSL (ADSL, HDSL, VDSL) digital subscriber lines.

The governing Maxwell's equations can be used to explain almost all the wave propagation mechanisms, from free space microwave propagation to antenna problems and optical fibre waveguides. This will be discussed in Chapter 4, Chapter 5 and Chapter 6.

Besides the technical impairments like attenuations, thermal noise, interferences and non-linear distortion, security issues also become more and more crucial and important. Since the internet is an open infrastructure enabling communications between people all over the world, malicious persons will misuse this opportunity to eavesdrop, manipulate confidential data, defraud, steal, blackmail and benefit financially from their criminal activities. Countermeasures both for the network operations of enterprises and private persons are most important in order to avoid any loss of confidential data and to defend themselves from any criminal activities. Therefore the network security and network management issues will be important parts of the information and communication technologies and networks, as discussed in Chapter 7.

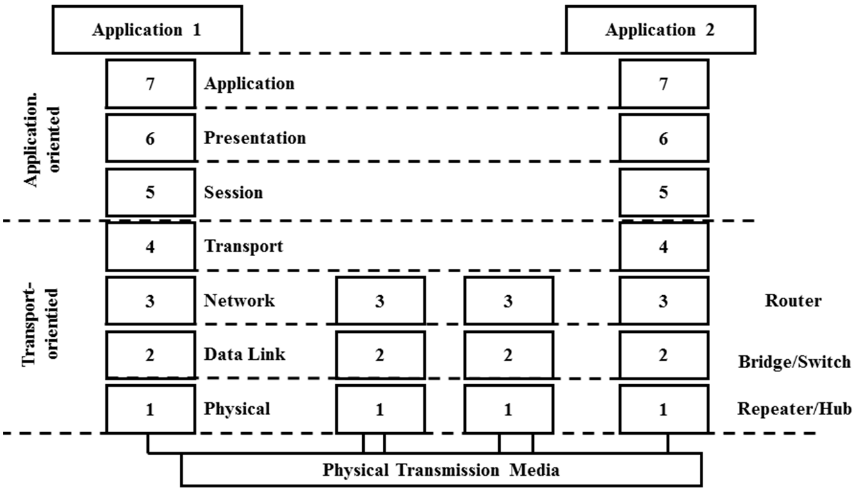


Figure 2 Open system interconnection model OSI

Open System Interconnection OSI model (Fig. 2) defines the communication standards. It defines seven layers to be treated separately and efficiently on the one hand, while on the other hand it allows that systems designed by different system suppliers or vendors, in accordance with the common OSI standards and other relevant technical standards like ITU, ETSI, IEEE and IETF, can be simply interconnected with each other flexibly and reliably. This OSI will be also the foundation for many network aspects.

Roughly speaking, the different layers fulfil different tasks and interact as a kind of master and slave to each other, top-down directly with the underlying layers:

- Application Layer: To allow access to network resources.
- Presentation Layer: To translate, encrypt and compress data.
- Session Layer: To establish, manage and terminate sessions.
- Transport Layer: To provide reliable process-to-process message delivery and error recovery.
- Network Layer: To move packets from source to destination and to provide internetworking.
- Data Link Layer: To organise bits into frames and to provide hop-to-hop delivery.
- Physical Layer: To transmit bits over a medium according to electrical and mechanical interface specifications.

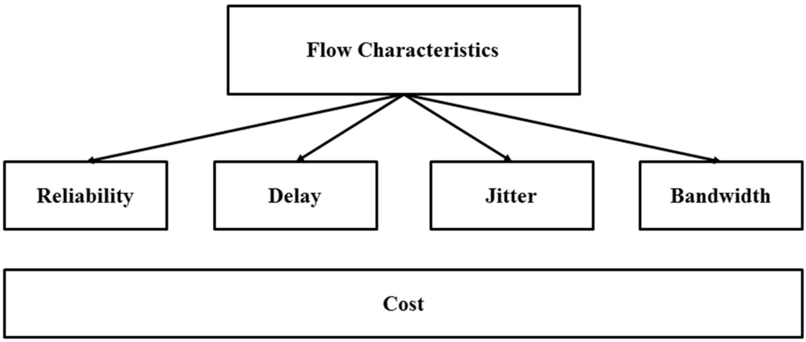


Figure 3 QoS parameters for the flow characteristics

ITU-T-defined QoS (quality of service) parameter sets, including service support, service operability, service accessibility, service retainability, service integrity, maintainability, availability, maintenance support, etc., were created to guarantee the reliable, timely and secure transmission of information with lowest bit error rates and low delay. In this book we will focus on the major technical parameters for the flow characteristics, i.e. availability or reliability, delay, jitter and available bandwidth (Fig. 3).

In fact, the QoS requirements for different services could be different. For example, voice or conversational video services are very sensitive to delay, but less sensitive to bit errors, whereas a data file transfer is extremely sensitive to bit errors, but less or even not at all sensitive to delay.

In order to provide reliable and secure communication between transmitter and receiver, the communication networks must be managed in a perfectly organised way. This is standardised by the network management systems TMN (Telecommunications Management Networks) and SNMP (Simple Network Management Protocol).

Throughout this book these terms of the communication channel model (Fig. 1) and the OSI model (Fig. 2) will be used often. At the end of this book, as outlook, future developments and trends will be discussed.

CHAPTER 2

INFORMATION THEORY

2.1 Shannon Capacity

A digital signal can be transmitted in a channel defined by its bandwidth (in Hertz) and by the noise and unwanted interference from other users. Since the total available frequency spectrum is limited, we are interested in the maximum data rate that can be transmitted over a channel with limited bandwidth and an existing noise and interference level.

In principle the data rate (in bps or bit/s) can be increased by increasing the number of bits per time. For simple modulation schemes, such as binary phase shift keying (see detailed explanation in Chapter 4), more bits/s consequently will increase the frequency spectrum of the signal and thus will require a larger bandwidth. Other modulation schemes such as higher order QAM (see Chapter 4) put the information bits, not only by using the phase shift, but also into amplitude steps. Obviously these systems require lower noise and interference within the channel, so that the receiver can reconstruct the original transmitted signal with its fine amplitude steps. Thus a higher data rate can be obtained with lower noise and interference, or a higher signal-to-noise ratio. The possible maximum data rate is called the channel capacity.

Claude Shannon published the first channel capacity estimation for digital data transmission stating that the total maximum channel capacity, or maximum data rate, depends basically on the available bandwidth B and the signal-to-noise ratio SNR.

$$C = B \cdot \log_2 \left(1 + \frac{S}{N} \right). \tag{2.1}$$

The signal noise ratio S/N can be also written as SNR [1]. This formula represents the maximum limit of the channel capacity C depending on the available spectrum or bandwidth B and the signal-to-noise ratio SNR. To be more precise, the SNR normally considers the ratio of the signal power level to all disturbing factors like noise and interferences. In the general case of wireline or wireless transmissions, not only the distance-dependant reduction of signal power *S* during the wave propagation in the air, the thermal noises *N* of transmitter and receiver circuits, as well as the transmission medium, but also the interferences *I* from the neighbouring transmitters in the same frequency band, will reduce the signal-to-noise ratio. In some papers this term is also written as $SINR = S/(I+N)$, so that the Shannon capacity formula can be written also as

$$C = B \cdot \log_2\left(1 + \frac{S}{I+N}\right). \tag{2.2}$$

Even though Shannon did not explicitly explain how to achieve the best channel capacity, this formula is the best approach to estimate the maximum channel capacity limit. In fact, all researchers working towards the goal to achieve the maximum capacity, to approach this maximum channel capacity limit, use sophisticated coding and modulation schemes as well as forward error correction techniques to exploit the given channel conditions.

Example:
Given the radio frequency bandwidth of 5 MHz and the signal-to-noise ratio of the wireless wave propagation environment SINR of 40 dB, what is the maximum channel capacity one can expect?

Hint: $$SINR_{dB} = 10 \cdot \log_{10}(SINR)\,. \tag{2.3}$$

Hint: $$SINR = 10^{(SINR_{dB}/10)}\,. \tag{2.4}$$

Example:
Given the available bandwidth of 10 THz for an optical fibre and the signal-to-noise ratio SINR (allowing for the attenuation loss of the optical wave propagation) to be 25 dB, what is the maximum channel capacity one can expect?

For the modern information and communication technologies, the information is transmitted as either an electrical or optical signal in digital modulation formats, as will be discussed in later chapters. Generally all information sources or parameters in real life (languages, music, images, pressure, velocity, acceleration, brightness, temperature, glucose, odour, etc.) can be converted to electrical and/or optical signals by using analogue digital converters. In the case of analogue amplitude modulation, the strength of these signals corresponds to the strength of the analogue electrical signals (current in Ampere, voltage in Volt, power in Watt). In digital modulation, the amplitudes will be converted into certain codes, which consist of only binary bits 0 and 1.

Coming back to the analogue amplitudes, if the signal levels have a very large dynamic range, or change strongly from very weak to very strong signals, it is not very practical to use the power levels between the output power level P_{out}= 0.000001 W or 1 μW and the input power level P_{in}=1 W. Instead we can use the logarithm to comprehensively represent the ratios of the power levels in dB. In this example the ratio of the input power level to the output power level will be then

$$a = \frac{P_{in}}{P_{out}} = \frac{1}{0.000001} = 1000000 = 10^6 . \tag{2.5}$$

As we will discuss in the next chapters, the electrical analogue signal is defined by the signal strength. Taking the power P we can define a ratio of the power at the input or power at the output of an arbitrary complex circuit in dB

$$a = 10 \cdot \log\left[\frac{P_{in}}{P_{out}}\right] = 10 \cdot \log\left[\frac{1}{0.000001}\right] = 10 \cdot \log\left[10^6\right] = 60 \text{ dB}. \tag{2.6}$$

We can also define the relative power level with a reference to 1 mW as dBm (or dBmW), or with reference to W as dBW for the power level at input or output, which makes the values more compact

$$p_{in} = 10 \cdot \log\left[\frac{P_{in}}{1mW}\right] = 10 \cdot \log\left[\frac{1W}{0.001W}\right] = 10 \cdot \log\left[10^3\right] = 30 \text{ dBm}$$

$$p_{out} = 10 \cdot \log\left[\frac{P_{out}}{1mW}\right] = 10 \cdot \log\left[\frac{0.000001W}{0.001W}\right] = 10 \cdot \log\left[10^{-3}\right] = -30 \text{ dBm}$$

$$p_{in} = 10 \cdot \log\left[\frac{P_{in}}{1W}\right] = 10 \cdot \log\left[\frac{1W}{1W}\right] = 10 \cdot \log\left[10^0\right] = 0 \text{ dBW}$$

$$p_{out} = 10 \cdot \log\left[\frac{P_{out}}{1W}\right] = 10 \cdot \log\left[\frac{0.000001W}{1W}\right] = 10 \cdot \log\left[10^{-6}\right] = -60 \text{ dBW}.$$

If the output power is higher than the input power, the circuit behaves as an amplifier. In the other case, if the output power is lower than the input power, the power is reduced within the circuit by means such as absorption or scattering. Physically, absorption means also the conversion of electrical or optical power into thermal energy.

It is this part of the lost energy that acts as disturbing noise, which is called thermal noise, arising when atoms and molecules vibrate and the temperature is non-zero. The higher the temperature, the stronger the vibration of the atoms and molecules will be. This noise vanishes only when theoretically the temperature is absolutely zero Kelvin 0 K = -273 °C, since in this case all the atoms or molecules are frozen completely. The thermal noise is constant with frequency, which is a good approximation for frequencies much lower than $k \cdot T/h$ or approximately 6 THz at room temperature or for most of the cases which we investigate in the radio frequency mobile communication networks:

$$N = k \cdot T \cdot B \tag{2.7}$$

with $k = 1.38 \cdot 10^{-23}$ Ws/K as the Boltzmann constant and $h = 6.6 \cdot 10^{-34}$ Ws^2 as the Planck constant. Depending on the spectral bandwidth B the total noise can be calculated.

Example:
Assuming a bandwidth of 20 MHz at a temperature of 300 K, the noise can be calculated as following

$$N = k \cdot T \cdot B = 1.38\ 10^{-23} Ws / K \cdot 300K \cdot 20 \cdot 10^{6} Hz = 8.28 \cdot 10^{-14}\ \text{W}.$$

With the above-mentioned output power P_{out}= 0.000001 W, and neglecting all other noise sources of other involved electronic circuits and interferences coming from other systems, we obtain the signal-to-noise ratio

$$\frac{S}{N} = \frac{P_{out}}{N} = \frac{10^{-6} W}{8.28 \cdot 10^{-14} W} = 1.208 \cdot 10^{7}$$

$$SNR = 10 \cdot \log\left[1 + 1.208 \cdot 10^{7}\right] = 70.8\ \text{dB}.$$

This leads to an estimated total channel capacity, or maximally achievable bit rate with the unit bps (bit per second), kbps, Mbps or Gbps

$$C = B \cdot \log_2(1 + \frac{S}{N}) = 20 \cdot 10^{6} Hz \cdot \log_2(1 + 1.208 \cdot 10^{7}) = 470 \cdot 10^{6}\ \text{bps}.$$

2.2 Multimedia Data

The final goal is to achieve the maximum channel capacity correspondingly to certain quality of service QoS requirements like delay, jitter, available spectral bandwidth, bit error rates or availability.

Generally all multimedia information sources can be represented by the binary information unit bits and bytes, in order to be transmitted by the digital communication networks and technologies:

Text: In data communications text is represented as a bit pattern, a sequence of 0s or 1s. Different sets of bit patterns are designed to represent text symbols. Each set is called a code and the process of representing the symbols is called coding. Famous examples for the coding methods are: Unicode and ASCII code, among others.

Numbers: Numbers are directly coded or converted to binary numbers.

Audio: Audio recording or broadcasting of sound or music can be done when the continuous audio signals are sampled in certain time steps, quantised in the amplitude and then converted into digital signal formats. For improving the efficiency, some compression techniques can be used. One of the most famous methods is MP3.

Images: Images consist of a matrix of pixels (small dots and picture elements). The resolution depends on the pixel size. Each pixel is assigned to a bit pattern. Several methods are used to represent the colour images, e.g. RGB (Red, Green, Blue) and YCM (Yellow, Cyan, Magenta).

Video: Videos, motion pictures, flash animations or movies consist of a certain number of subsequent images, for example 25 images per second. So generally speaking, the video recording and compression basically corresponds to the optimised processing of images in the corresponding time unit.

Advantages of digital information processing and transmission in comparison with the classical analogue techniques are:

- Lower costs with no need for expensive high-quality analogue circuits and components.

- Improved channel efficiency by utilising compression and multiplex techniques.
- Less impact from semiconductor component tolerances and interferences.
- Accuracy and resolution of digital circuits can be arbitrarily high.
- High transmission quality.
- High reliability by using sophisticated channel coding and FEC techniques.
- Long lifetime of the storage capability.
- Continuous reduction of the size and weight of the processors by continuous development of microelectronics, integrated circuits and computer technologies.
- Reduction of the access time to the storage and the signal transit time in the processor.

Disadvantages:

- High circuit complexity.
- High spectrum demand. This disadvantage can be partially compensated for by powerful compression techniques.

Driven by the large-scale integration in the integrated circuit IC design, down to the sub-micrometer or nanometer structure, the efficiencies of the ICs have been continuously improved in the last decades. Depending on the number of the basic semiconductor components (diodes and transistors) for the signal processing, the ICs can be characterised in the following categories:

SSI (Small Scale Integration):	<	100 components
MSI (Medium Scale Integration):	<	1.000 components
LSI (Large Scale Integration):	<	10.000 components
VLSI (Very Large Scale Integration):	<	100.000 components
ULSI (Ultra Large Scale Integration):	<	1.000.000 components
SLSI (Super Large Scale Integration):	<	10.000.000 components
ELSI (Extra Large Scale integration):	<	100.000.000 components
GLSI (Giant large scale integration):	>	100.000.000 components

Examples of digital information processing and signal transmission: analogue telephone technology PSTN has been almost completely replaced by digital wired and wireless telephone technologies such as ISDN (Integrated Service Digital Network), xDSL (ADSL Asymmetrical Digital Subscriber Lines, VDSL Very high bit rate Digital Subscriber Lines), GSM (Global System of Mobile Communications), UMTS (Universal Mobile Telecommunication System), LTE (Long Term Evolution) and 5G mobile communication technologies. Meanwhile even the first generation wired telephone network ISDN is being replaced by the IP-based telephony, Voice over IP, or by the so-called All-IP network. IP is the abbreviation of Internet Protocol.

In audio technology, the classical analogue audio storage system like magnetic tapes have already been replaced by CD (Compact Disc) or MP3 (MPEG 1 and MPEG 2 Audio Layer III). MPEG stands for Moving Picture Experts Group, the standardisation body for digital audio and video signal processing.

Analogue video signals have been replaced by digital video formats like DVB-x (DVB-S for satellite video broadcasting, DVB-T for terrestrial video broadcasting and DVB-C for cable video broadcasting by using hybrid fiber coaxial broadcasting networks), MPEG-2, MPEG-4 and HDTV. One recent successful example is H.264 (MPEG-4/AVC Advanced Video Coding), or even H.265 (HEVC High Efficiency Video Coding).

Generally all information parameters like amplitudes, brightness, colours, etc. can be converted from the analogue form to the digital form in a series of binary information units 0 and 1. As an example we have the signal in Figure 4.

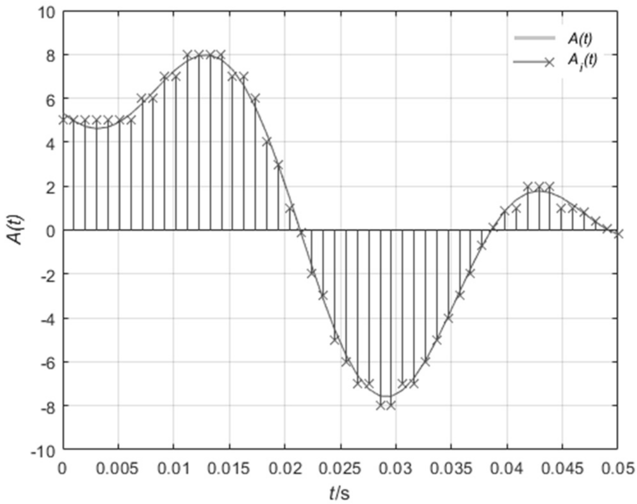


Figure 4 Sampling for analogue-digital conversion

The amplitude can be sampled with a period of ΔT, corresponding to a sampling frequency $f_{sampling}$, which is also called the sampling rate. Depending on the sampling rate, the temporal resolution or the sampling accuracy can be improved to an arbitrarily high value.

Shannon's sampling law (also known as Nyquist's criterion) requires the sampling frequency to be at least twice as high as the signal spectral bandwidth f_{Signal}, in order to fully represent the original analogue signal with no loss in information and to allow recovery of the original signal without deformation:

$$f_{Sampling} = \frac{1}{\Delta T} \geq 2 \cdot f_{Signal} \,. \tag{2.8}$$

By doing so, the continuous signal *A(t)* will be sampled to discrete signal values $A(t_i)$.

The discrete, sampled amplitudes $A(t_i)$ will further be quantised in 2^n-1steps or intervals, with the integer number *n*, which is the number of the bits used to represent or approximate the different amplitudes between the maximum amplitude A_{max} and minimum amplitude A_{min}, and with the step size

$$\Delta A = \frac{A_{\max} - A_{\min}}{2^n - 1} \,.$$

The discrete amplitudes are calculated by

$$A(i) = A(0) + \Delta A \cdot j \,.$$

By using this stepwise quantisation, the amplitude *A(t)* at the point in time t can be approximated by

$$A'(t_j) \cong A(j) \,, \text{ if } \; A(j) \geq A \geq A(j-1) + \frac{\Delta A}{2}$$

$$A'(t_j) \cong A(j-1) \,, \text{ if } \; A(j-1) + \frac{\Delta A}{2} \geq A(j) \geq A(j-1)$$

which corresponds to maximum quantisation error $\pm \Delta A / 2$

$$\left| A(t_j) - A'(t_j) \right| \leq \frac{\Delta A}{2} \,.$$

By doing so, all possible amplitudes between the allowable range of the maximum amplitude A_{max} and minimum amplitude A_{min} can then be approximated by $A'(t_j)$.

Again, the error for the amplitudes can be minimised to a wanted or required level depending on the choice of the number of the n bits to represent the amplitudes.

n bits	2^n-1 intervals	Error less than $\Delta A/2$	Relative to $(A_{max}$-$A_{min})$
1	1	$(A_{max}-A_{min})/(2^{(n+1)}-2) = (A_{max}-A_{min})/2$	*0.500*
2	3	$(A_{max}-A_{min})/(2^{(n+1)}-2) = (A_{max}-A_{min})/6$	*0.167*
3	7	$(A_{max}-A_{min})/(2^{(n+1)}-2) = (A_{max}-A_{min})/14$	*0.071*
4	15	$(A_{max}-A_{min})/(2^{(n+1)}-2) = (A_{max}-A_{min})/30$	*0.033*
5	30	$(A_{max}-A_{min})/(2^{(n+1)}-2) = (A_{max}-A_{min})60$	*0.017*
6	62	$(A_{max}-A_{min})/(2^{(n+1)}-2) = (A_{max}-A_{min})/124$	*0.008*
7	126	$(A_{max}-A_{min})/(2^{(n+1)}-2) = (A_{max}-A_{min})/252$	*0.004*
8	252	$(A_{max}-A_{min})/(2^{(n+1)}-2) = (A_{max}-A_{min})/504$	*0.002*

Table 1 Discretisation of the analogue signal for ADC

The amplitudes can then be described by a digital number or code, for example $A_{min} = A(0) \sim 0000$, $A_{max} = A(2^4) \sim 1111$, if $n = 4$ bits will be used, so that the analogue amplitudes as analogue numbers will be converted into a digital code, or series of binary bits, with a required accuracy for each amplitude value $A(t_j)$.

As we can see in the table above, the relative error in comparison with the maximum range of the possible amplitudes can also be minimised to a required, or theoretically an arbitrarily, low level, depending on the dynamic range of the amplitudes, albeit with increasing effort.

In the above example, the quantisation is linear and the step size ΔA is constant. In some cases the quantisation could also be non-linear, for example in case of voice signal quantisation for telephony. A-law is a companding algorithm for voice signal digital encoding used in Europe (in USA and Japan a similar μ-law is used), in order to modify the amplitude of the sampled voice signal, reduce the dynamic range and increase the signal-to-noise ratio. With A=87.6, the signal amplitude S is mapped to S′ (the signal amplitude S is chosen instead of A, in order to distinguish the A factor here):

$$S' = \frac{sign(S)}{\left(1+\ln(A)\right)} \cdot \begin{cases} A \cdot |S| & if\ |S| \le 1/A \\ 1+\ln(A \cdot |S|) & else \end{cases} . \qquad (2.9)$$

Example:
For the telephony application, the human voice can be bandpass-filtered in the spectrum from 300 Hz – 3400 Hz, without significantly degrading the voice quality and comprehensibility. By using the Shannon's sampling law (2.9) the sampling rate must be larger than

$$f_{Sampling} = \frac{1}{\Delta T} \ge 2 \cdot f_{Signal} = 6800 \text{ Hz}.$$

In fact, the sampling frequency or sampling rate is chosen with $f_{Sampling} = 8000$ Hz with a distance of 125 µs between the neighbouring samples, which will each be compressed to 8 bits representing the amplitude. At the end a data rate of 8 bits × 8000 Hz = 64 kbps for PCM encoding G.711 is achieved, which is used in the ISDN telephony.

A digital number is characterised by a certain number of bits. Each bit has only 2 states, either 0 or 1.

Examples for binary digital signals in electrical or optical information and communication technology:

- Electrical pulse is on or off.
- Electrical voltage is on or off.
- Optical transmitter, which is a laser diode, is switched on or off.
- Normally in electrical digital technologies, the 2 binary states are characterised by the voltage.
- For example the gates in the TTL logic present the logic states by the voltage level: 0V ("Low level", "L"), +5V ("High Level", "H").

In principle, the relation between the binary states L (Low) and H (High) to the logical states 0 and 1 could be arbitrary. Normally the so-called "Positive Logics" is applied, when:

The logic state "0" corresponds to the binary state "L" (low).
The logic state "1" corresponds to the binary state "H" (high).

The logic state "1" means in the logic design or Boolean logic "True" or "Correct", whereas the logic state "0" means in the logic design or Boolean logic "False" or "Wrong".

Of course, if required or meaningful for some applications, and, if the definition is used consequently, the "Negative Logics" can also be used, where the relation between the binary states L (Low) and H (High) to the logical states 1 and 0 are:

Logic state "1" corresponds to the binary state "L" (low).
Logic state "0" corresponds to the binary state "H" (high).

In this case, the logic state "0" means in the logic design or Boolean logic "True" or "Correct", whereas the logic state "1" means in the logic design or Boolean logic "False" or "Wrong". Normally the positive logic is used.

All these logic states can be controlled by the semiconductor integrated circuits based on the bipolar or MOSFET transistors and diodes.

2.3 Data Processing, Boolean Logics

The digital binary information of all multimedia data will be processed by Boolean logics. In the following, we first define the logic combinations "and", "or", "not" or negation. Values of the logic variables *A, B, C ... X, Y, Z* can be "0" or "1" [2].

$$Z = A \wedge B \quad \text{(AND)} \tag{2.10}$$
$$Z = A \vee B \quad \text{(OR)} \tag{2.11}$$
$$Z = \overline{A} = \neg A \quad \text{(NOT, negation)} \tag{2.12}$$

In Boolean algebra, these basic logics can also be combined:

$$Z = \overline{A \wedge B} = \overline{AB} \tag{2.13}$$
$$Z = \overline{A \vee B} \tag{2.14}$$
$$Z = (A \wedge \overline{B}) \vee (\overline{A} \wedge B) = A\overline{B} \vee \overline{A}B \tag{2.15}$$

AND	OR	NOT
$0 \wedge 0 = 0$	$0 \vee 0 = 0$	$\overline{1} = 0$
$0 \wedge 1 = 0$	$0 \vee 1 = 1$	$\overline{0} = 1$
$1 \wedge 0 = 0$	$1 \vee 0 = 1$	
$1 \wedge 1 = 1$	$1 \vee 1 = 1$	

Table 2 Boolean algebra

Similarly with the linear algebra, the Boolean algebra use also:

The Commutative laws:

$$A \wedge B \wedge C = C \wedge A \wedge B \tag{2.16}$$

$$A \vee B \vee C = C \vee A \vee B \tag{2.17}$$

The Associative laws:

$$A \wedge (B \wedge C) = (A \wedge B) \wedge C \tag{2.17}$$

$$A \vee (B \vee C) = (A \vee B) \vee C \tag{2.18}$$

The Distributive laws:

$$A \wedge (B \vee C) = (A \wedge B) \vee (A \wedge C) \tag{2.19}$$

$$A \vee (B \wedge C) = (A \vee B) \wedge (A \vee C) \tag{2.20}$$

The DeMorgan’s laws:

$$Z = \overline{A \wedge B} = \overline{A} \vee \overline{B} \tag{2.21}$$

$$Z = \overline{A \vee B} = \overline{A} \wedge \overline{B} \tag{2.22}$$

Shannon’s law of inversion:

$$\overline{f(A, B, C, ...; \vee, \wedge)} = f(\overline{A}, \overline{B}, \overline{C}, ...; \wedge, \vee) \tag{2.23}$$

The Absorption laws:

$$Z = A \wedge (\overline{A} \vee B) = A \wedge B \tag{2.24}$$

$$Z = A \vee (\overline{A} \wedge B) = A \vee B \tag{2.25}$$

$$Z = A \vee (A \wedge B) = A \tag{2.26}$$

$$Z = A \wedge (A \vee B) = A \tag{2.27}$$

By using these basic laws one can analyse logic circuits, or in a reverse way, design the logic circuit, in order to fulfil some tasks in engineering.

By using these basic Boolean logic or algebraic laws, computer chips, storage systems, sensors, analogue digital converters ADCs, digital analogue converters DACs, transmitters, receivers, source coders, channel coders and modulators can be built. Generally each of these information and communication networks, with a certain number of given input parameters A, B, C … etc., can be designed in order to process all the input data and to deliver a set of output parameters Z, Y, X …, etc.

The input parameters could be computer data, sensor data, user inputs, multimedia information to be transmitted to the receivers, and so on. The output data could correspondingly be machine control data or modulated digital data bit streams to be transmitted wirelessly or via wirelines.

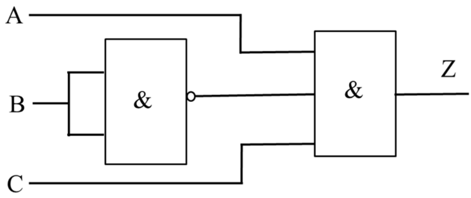
Example:
In order to demonstrate the logic design rules discussed above, a strongly simplified task is used [2] to design a digital circuit for safety purpose in such a way that the lift Z can lift up (Z=1) when, and only when:

1) The door contact sensor A shows that the lift door is closed (A=1). The other case is, the door is still open (A=0).
2) The overload sensor B shows that the lift is not overloaded (B=0). The other case is, the lift is overloaded B=1.
3) Someone pushes the button for one upper floor (C=1). If nobody pushes a button, then C=0.

This task can be translated into binary Boolean logic:

Case	C	B	A	Z
1	0	0	0	0
2	0	0	1	0
3	0	1	0	0
4	0	1	1	0
5	1	0	0	0
6	1	0	1	1
7	1	1	0	0
8	1	1	1	0

The control function Z for the lift will be then: $Z = A \wedge \overline{B} \wedge C$ which can be implemented with the digital logic gates, which in turn could be – for different applications and solutions – IC (mass production, consumer electronics), ASIC (application specific integrated circuit) or flexible FPGA (field programmable gate array, especially important for experimental setups and smaller technical problems):



2.4 Information Content, Entropy

In section 2.1 and section 2.2 we discussed channel capacity and multimedia data. Related to these topics is the information content. A multimedia message has certain information content. We will now discuss this information content in a more abstract or mathematical way.

Taking into account the different message units or symbols $m_1, m_2 \ldots, m_k \ldots m_n$ with the corresponding probabilities of occurrence $p_1, p_2 \ldots p_k \ldots p_n$ [1] [3] as an example. The symbols could be the alphabet of a language, used to express all the information contents. Shannon defined the amount of information of a symbol m_k as I_k with the binary information unit, or binary digit ("bit"), in a very strict technical sense, namely reciprocally proportional to the dual logarithm of the probability of occurrence.

$$I_k = \log_2 \frac{1}{p_k} . \tag{2.28}$$

Therefore the amount of the information each symbol carries directly depends on the probability of the occurence of this symbol in the message, which should be representative or large enough so that the probabilities of all symbols should be a stable average value.

If M symbols or messages are equally likely with $M = 2^N$, and independent from each other, the probability of each symbol will be $p_k=1/M$. The information content of each symbol is then

$$I_k = \log_2 \frac{1}{p_k} = \log_2 M = N . \tag{2.29}$$

If however the probabilities of the occurence of the symbols are different, so a long sequence of L messages will consist of different symbols m_k, each with the information I_k, then these L messages contain therefore the total information content

$$I_{Total} = p_1 L \log_2 \frac{1}{p_1} + p_2 L \log_2 \frac{1}{p_2} + \ldots + p_k L \log_2 \frac{1}{p_k} + \ldots + p_M L \log_2 \frac{1}{p_M} . \tag{2.30}$$

The average amount of the information of these L messages will then be defined as entropy $H= I_{Total} / L$

$$H = p_1 \log_2 \frac{1}{p_1} + p_2 \log_2 \frac{1}{p_2} + \ldots + p_k \log_2 \frac{1}{p_k} + \ldots + p_M \log_2 \frac{1}{p_M}$$
$$= \sum_{k=1}^{M} p_k \log_2 \frac{1}{p_k} \tag{2.31}$$

The maximum entropy can be achieved, if the distribution of the probabilities of the letters is equal. In this case the entropy will be

$$H_0 = \sum_{k=1}^{M} p_k \log_2 \frac{1}{p_k} = M \cdot \log_2 \frac{1}{1/M} = M \cdot \log_2 M \ . \tag{2.32}$$

Normally the maximum entropy cannot be achieved due to the unequal distribution of the probabilities of occurences. The difference of the maximum entropy H_0 and the realistic entropy H is called redundancy R.

$$R = H_0 - H \ . \tag{2.33}$$

As we will discuss later, the redundancy of an alphabet in a language or a code system is indispensable to allow for automatic FEC. This is also our daily experience. If somebody does not speak absolutely clearly, or if the cellular mobile phone reception is not very good, you may not understand every word of the conversation partner, but you still understand, at least you can imagine, what he means or what he could possibly say. The same is valid for written text. If someone writes a sentence with some missing or wrong letters, in many cases you can still understand the content of the text. This is exactly because each language has a lot of redundancy in the words and sentences, which are very helpful for the automatic error correction in daily communications. In information and communication technologies, redundancy is used to recognise and correct errors automatically in the similar way.

Taking the example of the English or German alphabet, the probabilities of different letters are totally different, leading to different amounts of information of each letter in an arbitrarily long text.