FUNDAMENTALS OF COMMUNICATION SYSTEMS

Przemysław Dymarski and **Sławomir Kula**
Warsaw University of Technology, Poland

Keywords: Analog and digital transmission systems, modulation, communication channel, source coding, channel coding, channel capacity, multiplexing, synchronization, carrier synchronization, symbol synchronization, clock synchronization, frame synchronization, packet synchronization, bit synchronization, multimedia synchronization, network synchronization

Contents

Summary

General presentation of the communication systems is given, including their classification, basic functions, performance measures and limits. Generic scheme of the analog and digital system is described and the advantages of the digital transmission are discussed. Problems of analog to digital conversion, lossless and lossy compression and source coding are briefly outlined. The channel codes and modulations are described and performance measures of the transmission system are presented. The impact of the limited capacity of the channel to the system performance is discussed. Special attention has been paid to the synchronization problems of signals and networks.

1. Introduction

Communication consists in transferring information in various form (text, audio, video) from one place to the other. This process needs at least three elements: an information source, a medium carrying information to a remote point and a sink collecting the transmitted information. We usually imagine that people are directly involved, at least as sources and addressees of the information, but nowadays it is not always the case: the equipment itself exchanges information e.g. remote control data. In the simplest case a messenger may transmit the information but usually some technical means are used for it. Therefore, a *communication system* may be defined as a set of facilities making possible the communication by means of signals. The last remark is necessary in order to exclude delivery of material objects e.g. letters. Post offices form a special kind of system, but discussion of its activities is out of the scope of this text.

Signals representing the information may be of an *analog* or *digital* form. Speech (represented by the varying acoustical pressure or continuously varying voltage) is an example of analog signal, text (e.g. represented by ASCII) is a digital signal. Historically the transmission of digital signals preceded the analog transmission: Samuel Morse invented his telegraph in 1837 but Graham Bell proposed his telephone in 1875. In the same way a digital wireless transmission preceded the analog one: Guglielmo Marconi initiated wireless communication over the English Channel in 1899 using electromagnetic waves caused by sparks jumping between two conductors, wireless transmission of the analog signals appeared later, with the invention of a triode. Soon the analog telephony and broadcast rapidly developed, but in the fifties of 20th century a comeback of digital transmission was noted. This process is at its final stage nowadays – telephony is almost entirely digital, audio and TV broadcast gradually evolves towards digital transmission. If the original signals appear in the analog form, the analog-to-digital (A/D) conversion and source coding is necessary, transforming these signals into a bit stream – series of logical values, zeros and ones. Further comments upon these techniques will be given in Section 2.

Why the transmission of digital signals is preferred to the transmission of analog signals? First of all, it is due to the immunity to noise. Transmitting medium (e.g. a cable) attenuates the signal and, in the case of analog transmission, regularly spaced amplifiers must be applied in order to keep the signal power above the noise level. In this way noise (e.g. thermal noise), introduced at the early stages, is amplified with the signal and gets accumulated. This process yields a long distance analog transmission not feasible. In the case of digital transmission the repertoire of signals being transmitted is finite (in the simplest case there are only two elementary signals - representing the logical zero and one). Therefore, instead of the amplifiers, the regenerative repeaters are used, which do not simply amplify the signal (and noise), but rather recognize which signal is being transmitted. If this recognition is proper (i.e. errorless), the signal is regenerated in its original form, without any noise. This description is simplified – in digital transmission there are some factors causing problems in a long distance transmission, e.g. difficulty in localizing the beginning and the end of each elementary signal being transmitted in presence of noise (i.e. problem of symbol synchronization).

There are the other advantages of digital transmission. Fiber optics, yielding low-cost and high capacity communication links, is more adapted to digital than analog transmission (e.g. it is easier to detect a presence of light than properly measure its intensity). Digital data assure high degree of security, due to cryptographic techniques, but the analog scrambling techniques (e.g. applied to speech signal) are quite easy to "pirate".

Communication systems use many transmission media, but each medium is usually shared by many signals carrying different information at the same time. This is possible due to *multiplexing* techniques. They were introduced in 1874 by Emile Baudot, who has installed a rotary switch, synchronized by the electrical signal, enabling to connect 6 telegraph operators to the same line. The transmission line could carry up to 90 bits per second in his system. Today this approach is called a *Time Division Multiplexing* (TDM), because each user obtains distinct time slots to transmit the information. It may use the same band (frequency range) as the other users. Though the concept of TDM dates from the 19th century, its worldwide career started much later. In the early 1960s in the USA, the TDM technique was applied in the 24-channel T1 carrier system. A few years later this technique was employed in the European PCM-32 transmission system. Presently this technique is widely used in SDH transmission systems, mobile communication systems and many others.

On the other hand, the *Frequency Division Multiplexing* (FDM) consists in attributing different frequency bands to the transmitted (at the same time) signals. An example is a commercial radio, where listeners must tune in to select a proper band. In the middle of 1920s, the FDM technique found a practical application in a transmission system operated on twisted pair line, afterwards it was implemented in wireless transmission, but a commercial success came with using FDM to carry multiple signals over coaxial cable. Presently, FDM technique is used for example in mobile telephone systems, in subscriber line system, and in cable TV.

In the WDM (*Wavelength Division Multiplexing*), the FDM concept is applied to the transmission using the optical fibers. Each of WDM optical channels operates at a different optical wavelength λ_i. Because length λ_i and frequency f_i of any wave are related by the formula

$$\lambda_i = \frac{v}{f_i}$$

where v is a propagation speed of the wave, the WDM technique formally could be considered as the FDM, but the WDM is applied in conjunction with fiber transmission while the FDM found its application either in wireless transmission systems or in metallic cables. Fiber optic transmission makes use of three, so called optical windows - the first around 850 nm, the second around 1300 nm and the third around 1550 nm. Today the 3rd window is commonly used in long distance transmission. Light emitted by a diode or laser is a narrowband signal. Width of light signal spectrum depends on emitter features, and varies from << 1 nm for SLM laser, 3 nm for MLM laser, to > 100 nm for LED diode. Because of it, if we want to transmit more than one light wave using the same fiber, neighboring wavelengths have to be separated by an unused band to prevent interfering. The most common spacing is about 100 GHz. In a case of so called DWDM (*Dense Wavelength Division Multiplexing*) smaller spacing values 50 GHz, 25 GHz, and even 12.5 GHz are used. Presently several hundred of wavelengths can be simultaneously transmitted over one fiber, so a maximum total bit rate of the DWDM transmission system reaches several Tbps.

In the *Code Division Multiplexing* (CDM) signals appear in the same time and use the same frequency band. Nevertheless it is possible to separate them, because each signal is orthogonal with respect to the others and a projection of the received sum of signals onto a proper direction in signal space yields the desired information and sets to zero the unnecessary information. CDM technique is used as an access technology, namely CDMA (*Code Division Multiple Access*), in UMTS (*Universal Mobile Telecommunications System*) standard for the third generation mobile communication. Another important application of the CDMA is GPS (*Global Positioning System*).

Classification of communication systems depends on criteria being used. For transmission of the analog signals, without using the A/D conversion, the *analog communication systems* are applied. For transmission of a bit stream (which may represent the original digital information, e.g. ASCII string, or the analog signal, after its conversion to a digital form), *digital communication systems* are used. The basic blocks of an analog communication system are given in Figure 1.

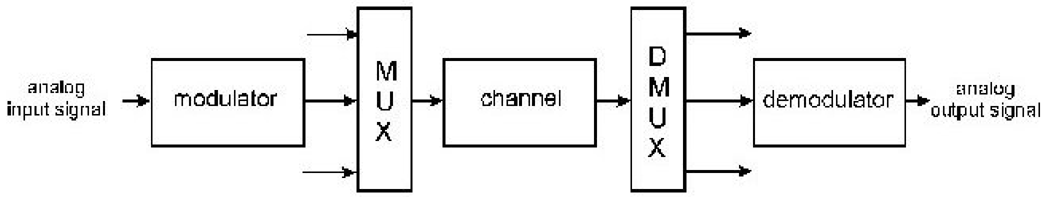

Figure 1: The analog communication system

The basic function of this system consists in adapting the properties of the analog signal carrying information (called the modulating signal) to the properties of a transmission channel (medium). The modulating signals (speech, audio, TV) have rather a baseband character – they occupy frequency band from (almost) 0 Hz to the maximum frequency (i.e. bandwidth) f_{M}. On the other hand, the transmission channel has a passband character, for example, commercial radio stations use some band with bandwidth B, centered at the carrier frequency f_0. The analog modulator, using one of known modulation techniques (AM – *amplitude modulation*, FM – *frequency modulation*, PM – *phase modulation*), pushes the spectrum of the modulating signal towards the high frequencies, thus making possible to pass the modulated signal through the channel. Generally a bandwidth is increased in this process (i.e. $B > f_{\mathrm{M}}$), but it has some advantage: the immunity to the channel noise also increases (this paradox shall be discussed in Section 4).

Several signals may be modulated using different channel subbands, thus making use of the FDM technique. At the receiver's side, demultiplexing is performed, using bandpass filters. Then a demodulator, performing an inverse operation with respect to a modulator, yields a demodulated signal. Due to distortions introduced in the channel (e.g. an additive noise) this signal is not an exact copy of the modulating signal.

In Figure 1 a *network* aspect of the analog communication is omitted. The role of network consists mainly in preparing a direct link between a source and a sink of signal. In a case of the analog telephony, in a telephone exchange (central office) cables are connected by means of switches, subbands are attributed in a frequency division multiplexer – thus yielding a circuit being unchanged during a call. This switching technique is called *circuit switching*. The switching techniques are described in *Analog and Digital Switching* and the problems concerning networks are discussed in

Fundamentals of Telecommunication Networks.

The digital communication system is presented in Figure 2.

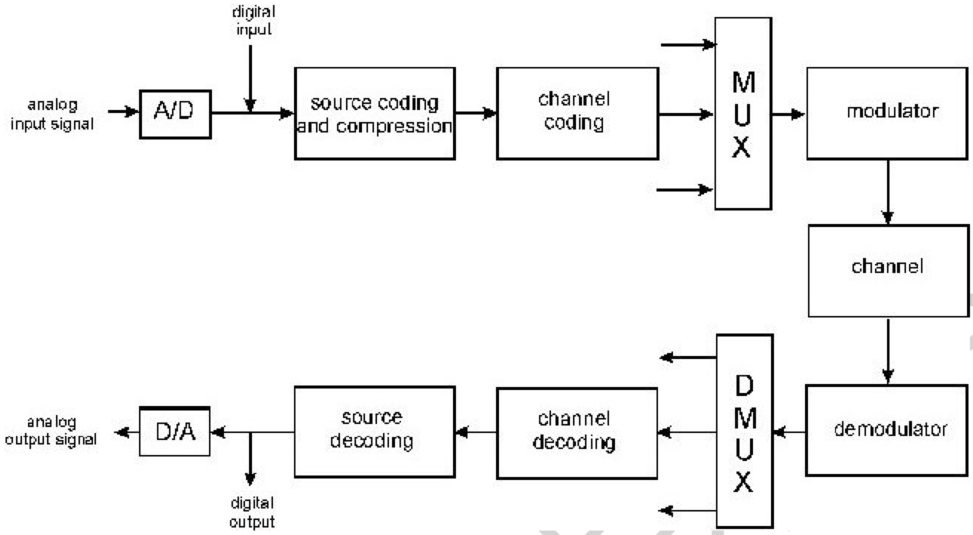

Figure 2: The digital communication system

If the input information appears in a form of an analog signal, it must be converted to a bit stream using a proper A/D converter. In this process some part of the information is lost – there exists an infinite number of analog signals of finite duration, but only a finite number of bit streams of finite length. The A/D conversion is sometimes performed in the moment of the acquisition of the analog information, e.g. in a digital video camera. Producers of the digital audio/video equipment tend to make this information loss perceptually insignificant, so they deliver a bit stream of a high rate (expressed in bits/s).

The bit rate may be decreased using *compression* algorithms based on *source coding* theory. The lossless compression yields a bit stream of reduced bit rate, which contains the same information as the initial bit stream. It is possible because of the redundancy of the information source – Samuel Morse used it in his alphabet proposed in thirties of the 19th century, coding frequently appearing letters in short code words (e.g. A=dot and dash, E=dot) and rarely appearing letters in long code words (e.g. Q=dash, dash, dot and dash). Further reduction of bit rate may be accomplished using lossy compression. In this process some information is lost and the original bit stream cannot be reproduced without errors. This is not acceptable if the original bit stream represents e.g. text data, but in the case of speech or video sequence some difference between the input and the output signal (i.e. the quantization error) may be accepted. This will be discussed in Section 2. At the source coding stage the encryption process may be applied, yielding the secure bit stream, which may be decoded using a secret key at the decryption stage.

At the channel coding stage the extra bits are appended to the bit stream, in order to detect and correct transmission errors. In this way, redundancy is increased, but the bit error rate (BER) is decreased (or, at least, the system is informed about the error and may react appropriately, e.g. it may demand a repetition of an erroneous codeword). The most elementary channel coder adds a parity control bit to each codeword, thus making possible to detect single errors within the codeword.

At this stage it is possible to mix bit streams coming from different sources using the TDM technique - thus forming a composite bit stream. Till now the set of logical values (zero and one) has been processed, but the channel transmits waveforms, which must be adapted to the properties of transmission medium. E.g. the optical fiber accepts light pulses, the radio transmission – signals based on a sinusoidal carrier. Thus a process of *modulation* is necessary, attributing elementary signals to the logical values or their strings. The simplest binary modulation attributes a signal $s_0(t)$ to the logical zero and $s_1(t)$ to the logical one. Waveforms $s_0(t)$, $s_1(t)$ must be adapted to the channel – their spectra must fit to the transfer function of the channel. In the case of baseband transmission spectra of the transmitted signals occupy the low frequency band and the corresponding modulation is called the *baseband modulation*, *line coding* or *transmission coding*. In the case of *M*-ary modulation there are *M* different signals (called symbols), each one carrying $\log_2 M$ binary logical values. The modulation and channel coding may be linked together, yielding a very efficient way of transmission – the trellis coded modulation (TCM).

Modulator may mix many bit streams using the FDM or CDM techniques. CDM requires orthogonalization of signals coming from one source with respect to signals coming from the other sources. This process increases bandwidth, so it is called a *spread spectrum* technique. It is advantageous from the point of view of immunity to distortions and interference introduced in the channel.

In a digital network the circuit switching technique has been used, but now the other technique appeared – the *packet switching*. The bit stream is cut into packets, usually of the same length, and the address is attributed to each packet. Network directs packets to the destination point, using the available means. No physical link is created between the sender and the addressee – packets of the same bit stream may use different routes to the sink.

At the receiver side, the transmitted elementary signals (symbols) must be recognized with a minimum number of errors. This is not possible without the synchronization at the symbol level - the decision instants must be aligned with time slots attributed to the elementary signals (Section 3.4). The proper recognition is difficult in presence of channel noise and the other distortions (fading, interference, etc.). Some packets may be lost or delayed in the network so they cannot be used in reconstruction of the original signal.

The received symbols are then transformed to a bit stream, the error detection and correction mechanism is applied, then, after decryption (if necessary) and decoding of the source code the output bit stream is directed to a digital sink. It may be used to reproduce (with finite precision) the analog signal carrying the information being transmitted.

The more detailed description and comparison of the analog and digital transmission systems may be found in *Analog and Digital Transmission*.

Different criteria may be used for classification of communication systems. There are systems enabling *duplex* communication (in two directions simultaneously, e.g. telephony), *half-duplex* (signals flow in one direction at a time) and *simplex* (signals flow in one direction, e.g. TV, commercial radio broadcast).

If the transmission media are used as a criterion, there are systems using *guided* transmission (e.g. open cable, twisted pair, coaxial cable, optical fiber), and systems using *wireless* (terrestrial or satellite) transmission. Modern systems use many transmission media at a time, e.g. cellular systems use optical fibers and wireless transmission. Systems may be also classified according to the frequency range used for transmission: baseband, radio frequency, microwave, infrared, optical systems. According to the bandwidth there are narrowband and wideband transmission systems. Bandwidth depends on the transmission medium: the open cables offer much smaller bandwidth than e.g. the coaxial cables.

Taking into consideration synchronization aspects, we have *asynchronous* systems, *plesiochronous* systems and *synchronous* systems (see Section 3.4).

The cellular communication systems are classified according to the technical parameters and technology applied. Thus there were the first generation (analog) systems, the second generation (2G) systems like the GSM, offering basically the digital voice communication links, the third generation (3G) systems like the UMTS, processing also the interactive multimedia like teleconferencing and Internet access at the rates up to 2 Mb/s, and the future 4G systems, yielding high data rates up to 20 Mb/s. See *Wireless Terrestrial Communications: Cellular Telephony* for more details.

2. Sources of Information and Source Coding

The modern digital communication systems process many kinds of signals carrying information. Some of them appear in an analog form and must be converted to a bit stream. In the case of wideband systems using optical fibers the resulting bit rate (number of bits per second) is not very critical. In the case of wireless transmission, e.g. cellular telephony, the bandwidth is limited and the bit rate must be reduced as much as possible. In the Table1 some examples of the input signals are given, with the available bit rates of the compressed bit stream. Note a wide range of bit rates, indicating that there is always a tradeoff between signal quality and the bit rate.

signal carrying information	bit rate (after compression)
telephonic speech (300-3400 Hz)	0.8 – 64 kb/s
wideband speech (< 7 kHz)	16 – 64 kb/s
wideband audio (< 22 kHz)	32 – 400 kb/s (mono)
videophone	16 – 64 kb/s
videostreaming (e.g. for UMTS)	50 – 150 kb/s
TV	1.5 – 15 Mb/s

Table 1: Examples of the input signals and their bit rates

There are two processes involved in A/D conversion: the discretization in time domain (*sampling*) and in amplitude domain (*quantization*). Sampling process may be, under certain conditions, fully reversible – no information is lost. According to the widely known Shannon sampling theorem, any signal $x(t)$ of a bandwidth f_{M} may be sampled at the sampling frequency greater than $2f_{\mathrm{M}}$ without any loss of information. This theorem stems from the Whittaker – Shannon interpolation formula, expanding the band limited baseband signal in the orthogonal basis of sinc functions:

$$x(t)=\sum_n x_n \frac{\sin\pi\left(t-nT\right)/T}{\pi\left(t-nT\right)/T}$$

where $x_n = x(nT)$ and T is a sampling interval, the inverse of the sampling frequency. The above formula states that the continuous analog signal $x(t)$ may be reconstructed, with an infinite precision, from its samples. The mechanism of this reconstruction is presented in Figure 3.

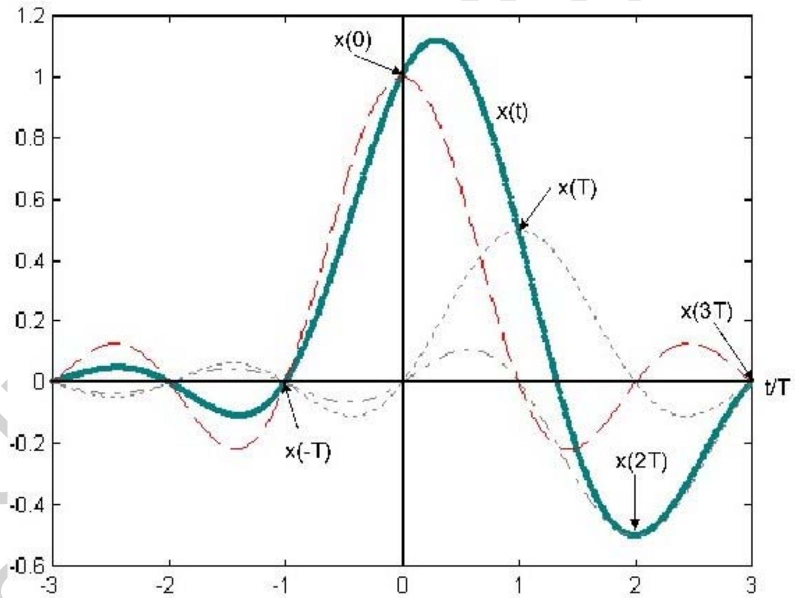

Figure 3: Reconstruction of the baseband analog signal from its samples

The quantization process is generally not reversible – the difference between the input signal samples (x_n) and the quantized samples $(x*_n)$ is called a quantization error $(e_n = x_n - x*_n)$. The quantizer tends to minimize the quantization error power $(\sigma_e^{\,2} = \mathrm{E}(e_n^{\,2})$, where E denotes the expected value) at the demanded resolution b bits per sample. Minimum of the error power yields maximum of the signal to noise ratio $SNR = \mathrm{E}(x_n^{\,2}) / \mathrm{E}(e_n^{\,2})$.

In the most simple case of the *scalar quantizer*, every signal sample is represented by one of the L quantization levels (the nearest one, in order to reduce the quantization error), and the coded number (index) of the chosen level is transmitted to the receiver. Different techniques are used to increase the *SNR*:

- Quantization levels are nonuniformly spaced, in order to reduce the quantization error power for signals of small amplitude (so called *logarithmic quantizer* used in the most popular *pulse code modulation* technique applied in digital telephony).
- Quantization levels are expanded or “shrunk”, in order to adapt the quantizer to the varying amplitude of signal being quantized (the *adaptive quantizer*).
- Each sample is predicted using the previous samples – the quantizer processes only unpredictable part of the sample and the predictable part is reconstructed at the receiver side using the same predictor as at the transmitter side (this technique is called *differential coding* or *adaptive differential pulse code modulation*).
- Adjacent samples of an audio signal or pixels of a picture are grouped in N-dimensional vectors – each vector is then represented by one of the L codebook vectors. This technique is called *vector quantization* – Figure 4.
- The codebook of the vector quantizer is filtered using an adaptive predictive filter, in order to adapt the codebook waveforms to the varying properties of speech signal (this is called a *Code Excited Linear Prediction* and is widely used in cellular telephony).
- Blocks of audio signal samples or adjacent pixels of a picture are transformed to frequency domain and then quantization is performed. This is called the *transform coding* and makes it possible to exploit the human perception in order to mask the quantization error.

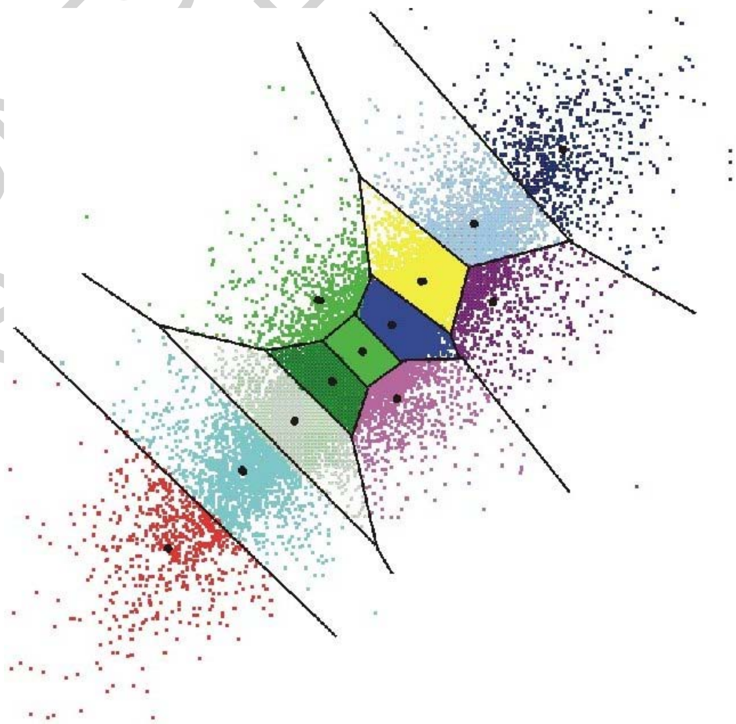

Figure 4: Vector quantization in N=2-dimensional space using a codebook of L=12 vectors

The success of the above mentioned techniques depends not only on the quantizer but also on the signal itself. According to the Shannon source coding theory for each signal and the resolution b it exists a lower bound for the coding distortion (e.g. for a quantization error power). It is given by the *Rate Distortion Function* (RDF). For the uncorrelated Gaussian signal the RDF is given by the formula:

$$\sigma_{e,\min}^2 = \sigma_x^2 \, 2^{-2b}$$

For a correlated signal lower distortions may be obtained (Figure 5), because the coding algorithm may use correlations e.g. for prediction of consecutive samples. The vector quantizer also uses correlations and other relations between N samples forming a vector. At a given resolution e.g. b=1 bit per sample, a scalar quantizer having $L=2^b$ quantization levels may be used, yielding a poor performance (a point N=1 in Figure 5). However at the same resolution a N-dimensional vector quantizer may be used, having $L=2^{bN}$ codebook vectors. Due to the exploitations of signal correlations and the other properties of the N-dimensional probability density function of the signal samples, distortion is reduced and gradually approaches (as N tends to infinity) the RDF limit. Thus the vector quantizer is an asymptotically optimal source coder. Processing of high-dimensional vectors causes a complexity problem: the number of codebook vectors increases rapidly. Therefore the other techniques are used together with the vector quantization (e.g. prediction and vector quantization form a basis of the code excited linear predictive speech coders).

Figure 5: The RDF of the uncorrelated and some correlated Gaussian signal and distortion obtained due to the scalar and N-dimensional vector quantization of this correlated signal at 1 bit per sample

-
-
-

Bibliography

Bregni S. (2002) Synchronization of Digital Telecommunications Networks. Wiley. [Overview of synchronization of networks and systems]

Haykin S. (2001). *Communication Systems*. Wiley. [A coherent system approach to the analysis of telecommunications]

Muller N.J. (2002). *Desktop Encyclopedia of Telecommunications.* McGraw-Hill. [Explanation of basic terms, addressed to the nontechnical professionals]

Smith R.D. (2003). *Digital Transmission Systems.* Kluwer. [Presentation of modern transmission systems]

Wozencraft J.M. Jacobs I.M. (1965,1990) *Principles of Communication Engineering* (Wiley) [Theoretical background to transmission, coding, channel capacity etc.]

Ziemer R.E., Peterson R.L. (2000) *Introduction to Digital Communication.* (Prentice-Hall). [Concepts of digital modulation, coding, spread spectrum etc.]

Biographical Sketches

Przemysław Dymarski received the M.Sc and Ph.D degrees in electrical engineering from the Wrocław University of Technology, Wrocław, Poland, in 1974 and 1983, respectively, and the D.Sc degree in telecommunications from the Faculty of Electronics and Information Technology of the Warsaw University of Technology, Warsaw, Poland, in 2004. Now he is with the Institute of Telecommunications, Warsaw University of Technology. His research includes various aspects of digital signal processing, particularly speech and audio compression for telecommunications and multimedia, text-to-speech synthesis and audio watermarking. Results of his research have been published in the IEEE Trans. on Speech and Audio Processing, IEEE Trans. on Audio, Speech and Language Processing, Annales de Télécommunications, and in the proceedings of international conferences like ICASSP, EUROSPEECH, EUSIPCO, etc.

Sławomir Kula received the M.Sc degree in electronic engineering from the Warsaw University of Technology, Poland, in 1977 and the Ph.D. degree in telecommunications from the Faculty of Electronics and Information Technology of the Warsaw University of Technology in 1982. Now he is a Vice Dean of that Faculty. His research includes all aspects of transmission systems and core, metropolitan and access networks, and signal (speech and video) processing. He gave lectures in the following countries: Algeria, Great Britain, France, Korea, Mexico and Poland. He was a senior expert of ITU-T, and he is an expert of PCA. He is the author of the book: Transmission Systems, edited by WKŁ (Warsaw, 2004) and coeditor and coauthor of the book: Synchronous Transmission Systems and Networks, also edited by WKŁ (Warsaw, 1996). Dr Kula is a member of IEEE ComSoc, a member of Communication Engineering Society (SIT) and the President of the SIT at Warsaw University of Technology.