Ασύρματα Δίκτυα Αισθητήρων

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Το φυσικό επίπεδο

- One of the desirable aspects of wireless sensor nodes is their ability to communicate over a wireless link.
- Because of it, mobile applications can be supported; flexible deployment of nodes is possible; and the nodes can be placed in areas that are otherwise inaccessible to wired nodes.
- Once the deployment is carried out, it is possible to rearrange node placement in order to attain optimal coverage and connectivity; and the rearrangement can be made without disrupting the normal operation of the structure or process the nodes monitor.
- However, wireless communication poses some formidable challenges. Some of these challenges are limited bandwidth, limited transmission range, and poor packet delivery performance because of interference, attenuation, and multipath scattering.
- In order to tackle these challenges, it is vital to understand their properties and some of the mitigation strategies that are already in place.
- This presentation provides a fundamental introduction to point-to-point wireless digital communication.

Το φυσικό επίπεδο

- The basic components of a digital communication system are the transmitter, the channel, and the receiver.
- Since wireless sensor nodes are placed close to each other in a wireless sensor network, here short range communication is of interest.
- For a more comprehensive treatment of wireless and digital communications, the reader is referred to Proakis (2000) and Wilson (1995).

Το φυσικό επίπεδο

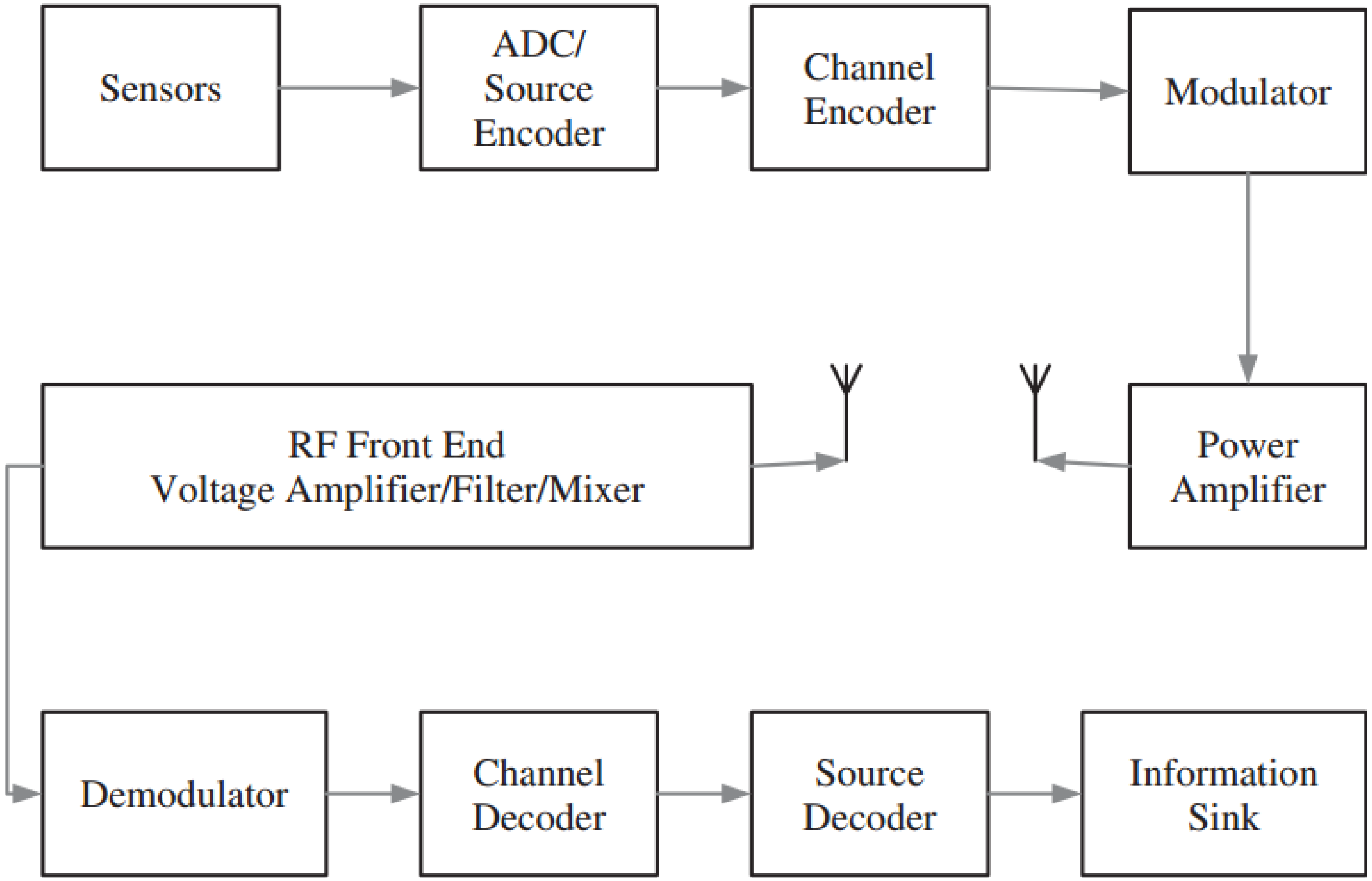

Figure 5.1 Components of a digital communication system.

Το φυσικό επίπεδο

- Figure 5.1 provides a block diagram of a digital communication system.
- The communication source in the context of this presentation represents one or more sensors and produces a message signal, an analog signal.
- The signal is a baseband signal having dominant frequency components near zero.
- The message signal has to be converted to a discrete signal (discrete both in time and amplitude) in order to be processed by the processor subsystem.
- The conversion requires sampling the signal at least at Nyquist rate, so that no information will be lost.
- After sampling, the discrete signal is converted to a binary stream. This process is called *source encoding*.
- It is essential to implement an efficient source-coding technique so that the channel's bandwidth and signal power requirements are satisfied

Το φυσικό επίπεδο

- One way to achieve this is by defining a probability model of the information source, so that the length of each information symbol depends on its probability of occurrence.
- The next step is channel encoding and its aim is to make the transmitted signal robust to noise and interference.
- Moreover, in case of signal corruption, it enables an error to be recognized and the original data to be recovered.
- There are two essential approaches: to transmit symbols from a predetermined codebook, and to transmit redundant symbols.

Το φυσικό επίπεδο

- After channel encoding, modulation takes place.
- This is a process by which the baseband signal is transformed into a bandpass signal.
- Modulation is useful for various reasons, but the main reason is to transmit and receive signals with short antennas.
- In general, the shorter the wavelength of the transmitted signal, the shorter is the length of the antenna.
- Finally, the modulated signal has to be amplified and the electrical energy is converted into electromagnetic energy (electromagnetic radiation) by the transmitter's antenna, and the signal is propagated over a wireless link to the desired destination.

Το φυσικό επίπεδο

- The components of the receiver block carry out the reverse process to retrieve the message signal from the electromagnetic waves.
- The receiver antenna induces a voltage that is, ideally, similar in shape, frequency, and phase with the modulated signal.
- Due to various types of losses and interferences, the magnitude and shape of the signal is changed and has to pass through a series of amplification and filtering processes.
- It is then transformed back to a baseband signal through the process of demodulation and detection.
- Finally, the baseband signal undergoes a pulse-shaping process and two additional stages of decoding (channel and source) in order to extract the sequence of symbols that represent the original analog signal, which is the message

Το φυσικό επίπεδο

- A source encoder transforms an analog signal into a digital sequence.
- The process consists of sampling, quantizing, and encoding.
- To explain these stages, suppose a sensor produces an analog signal that can be expressed as *s(t)*.
- During the sampling process, *s(t)* will be sampled and quantized by the analog-todigital converter (ADC) that has a resolution of *Q* distinct values.
- As a result, a sequence of samples, *S* = *(s*[1]*,s*[2]*,...,s*[*n*]*)* are produced.
- The difference between the sampled *s*[*j*] and its corresponding analog value at time *tj* is the quantization error.
- As the signal varies over time, the quantization error also varies and can be modeled as a random variable with a probability density function, *Ps(t)*.

Το φυσικό επίπεδο

- The aim of the source encoder is to map each quantized element, $s[j]$ into a corresponding binary symbol of length r from a codebook, C.
- If all the binary symbols in the codebook are of equal length, the codebook is called a Block Code.
- Often, however, the symbol length as well as the sampling rate is not uniform.
- It is customary, therefore, to assign short-sized symbols and high sampling rates to the most probable sample values and long-sized symbols and low sampling rates to less probable sample values.
- Figure 5.2 illustrates the input–output relationship of a source encoder.

Το φυσικό επίπεδο

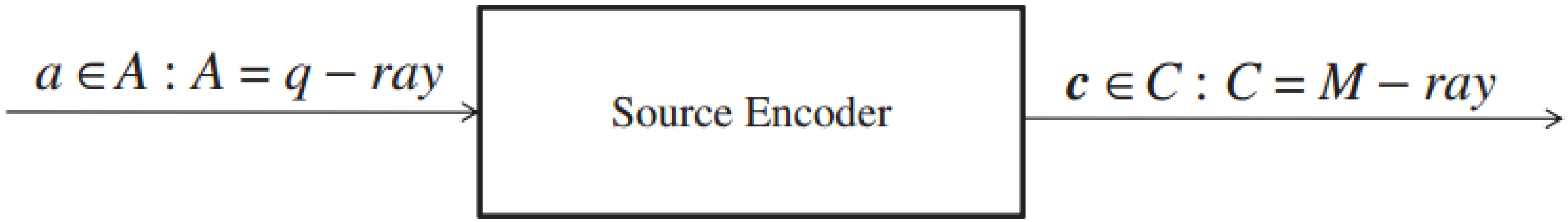

Figure 5.2 Input – output relationship of a source encoder.

Το φυσικό επίπεδο

- A codebook, *C*, can be uniquely decoded, if each sequence of symbols, *(C(*1*),C(*2*),...)* can be mapped back to a corresponding value in *S* = *(s*[1]*,s*[2]*,...,s*[*n*]*)*.
- A binary codebook has to satisfy Equation (5.1) to be uniquely decoded.

$$\sum_{i=1}^{u} \left(\frac{1}{r}\right)^{l_i} \leq 1 \tag{5.1}$$

- where *u* is the size of the codebook and *li* is the size of the codeword *C(i)*.

Το φυσικό επίπεδο

- A codebook can be instantaneously decoded if each symbol sequence can be extracted (decoded) from a stream of symbols without taking into consideration previously decoded symbols.
- This will be possible if and only if there does not exist a symbol in the codebook, such that the symbol **a** = $(a1, a2, ..., am)$ is not a prefix of the symbol **b** = $(b1, b2, ..., bn)$, where $m < n$ and $ai = bi$, $\forall i = 1, 2, ..., m$ within the same codebook.
- Table 5.1 lists different types of codebooks.

Το φυσικό επίπεδο

Table 5.1 Source-encoding techniques

	C^1	C^2	C^3	C^4	C^5	C^6
s_1	0	00	0	0	0	0
s_2	10	01	100	10	01	10
s_3	00	10	110	110	011	110
s_4	01	11	11	1110	111	111
Block code	No	Yes	No	No	No	No
Uniquely decoded	No	Yes	No	Yes	Yes	Yes
$\sum_{i=1}^{n} \left(\frac{1}{2}\right)^{l_i}$	$1\frac{1}{4}$	1	1	$\frac{15}{16} < 1$	1	1
Instantly decoded	No	Yes (block code)	No	Yes (comma code)	No	Yes

Το φυσικό επίπεδο

- The efficiency of a source encoder is a quantity that expresses the average length, *L(C)* = *E*[*li(C)*] of symbols used to represent the sampled analog signal.

- Suppose the probability of a *q*-ary source – that is, it has *q* distinct symbols – producing the symbol *si* is *Pi* and the symbol *Ci* in a codebook is used to encode *si* .
- The expected length of the codebook is given by:

$$L(C) = \sum_{i=1}^{q} P_i \cdot l_i(C) \tag{5.2}$$

Το φυσικό επίπεδο

- Sometimes, it is necessary to express efficiency in terms of the information entropy or Shannon’s entropy.
- In information theory, Shannon’s entropy is defined as the minimum message length necessary to communicate information.
- It is related to the uncertainty associated with the information. If the symbol *si* can be expressed by a binary symbol of *n* bits, the information content of *si* is:

$$I(s_i) = -\log_2 P_i = \log_2 \frac{1}{P_i}$$

Το φυσικό επίπεδο

- The entropy (in bits) of a *q*-ary memoryless source encoder is expressed as:

$$H_r(A) = E\left[I_r(s_i)\right] = \sum_{i=1}^{q} P(s_i) \cdot I_r(s_i) = \sum_{i=1}^{q} P(s_i) \cdot \log_2 \frac{1}{P(s_i)} \qquad (5.4)$$

- The efficiency of a source encoder in terms of entropy reveals the unnecessary redundancy in the encoding process. This can be expressed by:

$$\eta(C) = \frac{H(S)}{L(C)}$$

- The redundancy of the encoder is:

$$\frac{L - H(S)}{L} = 1 - \eta$$

Το φυσικό επίπεδο

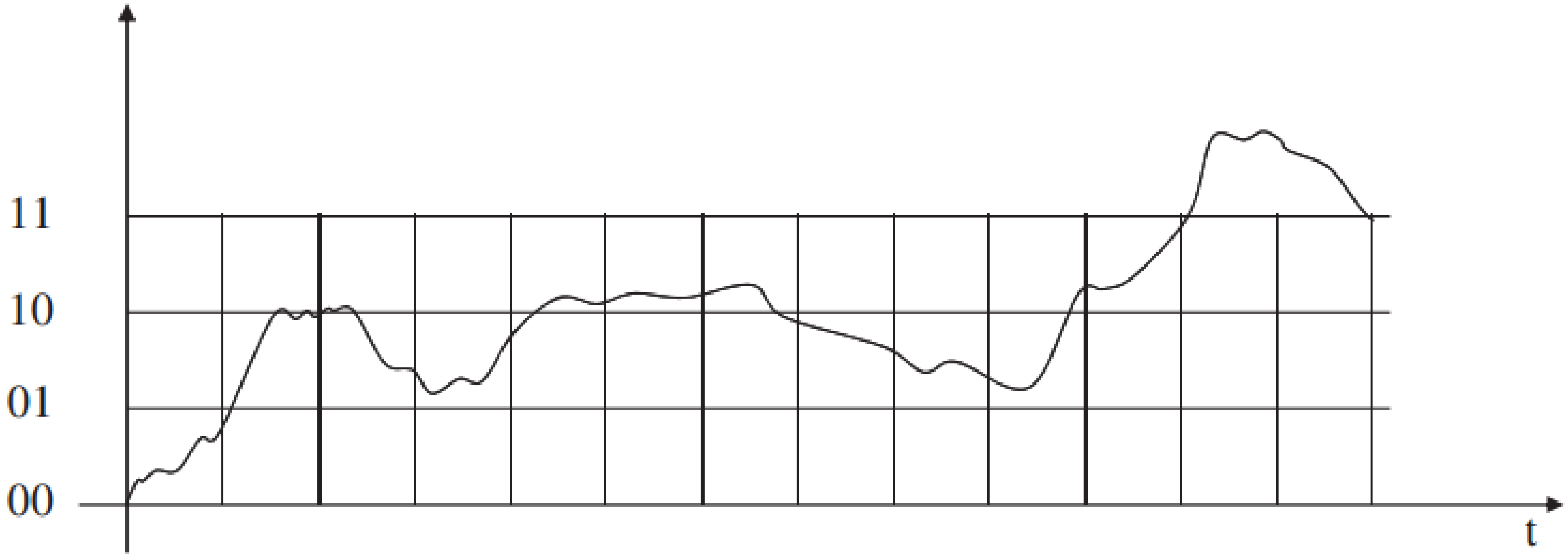

Figure 5.3 An analog signal with four possible values.

Το φυσικό επίπεδο

Suppose the analog signal in Figure 5.3 is quantized into four distinct values, 0, 1, 2, 3. As can be seen in the figure, some values (2) occur more frequently than others (0 and 3). If the probability of occurrence of these values can be expressed as $P(0) = 0.05$, $P(1) = 0.2$, $P(2) = 0.7$, $P(3) = 0.05$, then, it is possible to compute the efficiency of two of the codebooks given in Table 5.1, namely C^2 and C^3.

For $P_1 = 0.05$, $\log_2\left(\frac{1}{0.05}\right) = 4.3$. Because l_i has to be a whole number and there should be no loss of information, l_1 must be 5. Likewise, $l_2 = 3$; $l_3 = 1$; and $l_4 = 5$. Hence:

$$E[L(C^2)] = \sum_j l_j \cdot P_j = (5 \times 0.05) + (3 \times 0.2) + (1 \times 0.7) + (5 \times 0.05) = 1.8 \quad (5.7)$$

Το φυσικό επίπεδο

Using Equation (5.4), the entropy of C^2 is calculated as:

$$H(C^2) = 0.05\log_2\left(\frac{1}{0.05}\right) + 0.2\log_2\left(\frac{1}{0.2}\right) + 0.7\log_2\left(\frac{1}{0.7}\right) + 0.05\log_2\left(\frac{1}{0.05}\right) = 1.3 \tag{5.8}$$

Therefore, the encoding efficiency of the codebook, C^2 (see Table 5.2) is:

$$\eta(C^2) = \frac{1.3}{1.8} = 0.7 \tag{5.9}$$

The redundancy in C^2 is:

$$\mathrm{rdd}_{C^2} = 1 - \eta = 1 - 0.67 = 0.3 \tag{5.10}$$

Το φυσικό επίπεδο

In terms of energy efficiency, this implies that 30% of the transmitted bits are unnecessarily redundant, because C^2 is not compact enough.

In the same way l_j is computed for C^2, the expected symbol length (in bits) for C^3 (see Table 5.3) is given as:

$$\begin{aligned} E[L(C^3)] &= \sum_j l_j \cdot P_j \\ &= (3 \times 0.05) + (2 \times 0.2) + (1 \times 0.7) + (3 \times 0.05) \\ &= 1.4 \end{aligned} \tag{5.11}$$

Το φυσικό επίπεδο

Table 5.2 Description of the compactness of C^2

j	$\mathbf{a}_j$	P_j	l_j
1	00	0.05	5
2	01	0.2	3
3	10	0.7	1
4	11	0.05	5

Table 5.3 Description of the compactness of C^3

j	$\mathbf{a}_j$	P_j	l_j
1	100	0.05	3
2	11	0.2	2
3	0	0.7	1
4	110	0.05	3

Το φυσικό επίπεδο

Because the probabilities of the symbols are unchanged, entropy also remains unchanged. The encoding efficiency of C^3 is therefore:

$$\eta(C^3) = \frac{1.3}{1.4} = 0.9 \tag{5.12}$$

The redundancy, rdd, in C^3 is:

$$\mathrm{rdd}_{C^3} = 1 - \eta = 1 - 0.9 = 0.1 \tag{5.13}$$

Το φυσικό επίπεδο

- Pulse code modulation (PCM) and delta modulation (DM) are the two predominantly employed source encoding techniques.
- In digital pulse code modulation, the signal is first quantized and then each sample is represented by a binary word from a finite set of words.
- The size of the individual words as well as the number of words in the set determines the resolution of a PCM technique and the source encoder bit rate.
- In PCM information is conveyed in the presence or absence of pulses and not in the amplitude or the location of the edges of the pulses.
- Because of this property, PCM greatly enhances (almost noise free) the transmission and regeneration of binary words.
- The associated cost with this form of source encoding is the quantization error and the energy and
  bandwidth required to transmit multiple bits for each sampled output.
- Figure 5.4 illustrates a PCM technique that uses two bits to encode a single sample. Four distinct levels are permissible during sampling

Το φυσικό επίπεδο

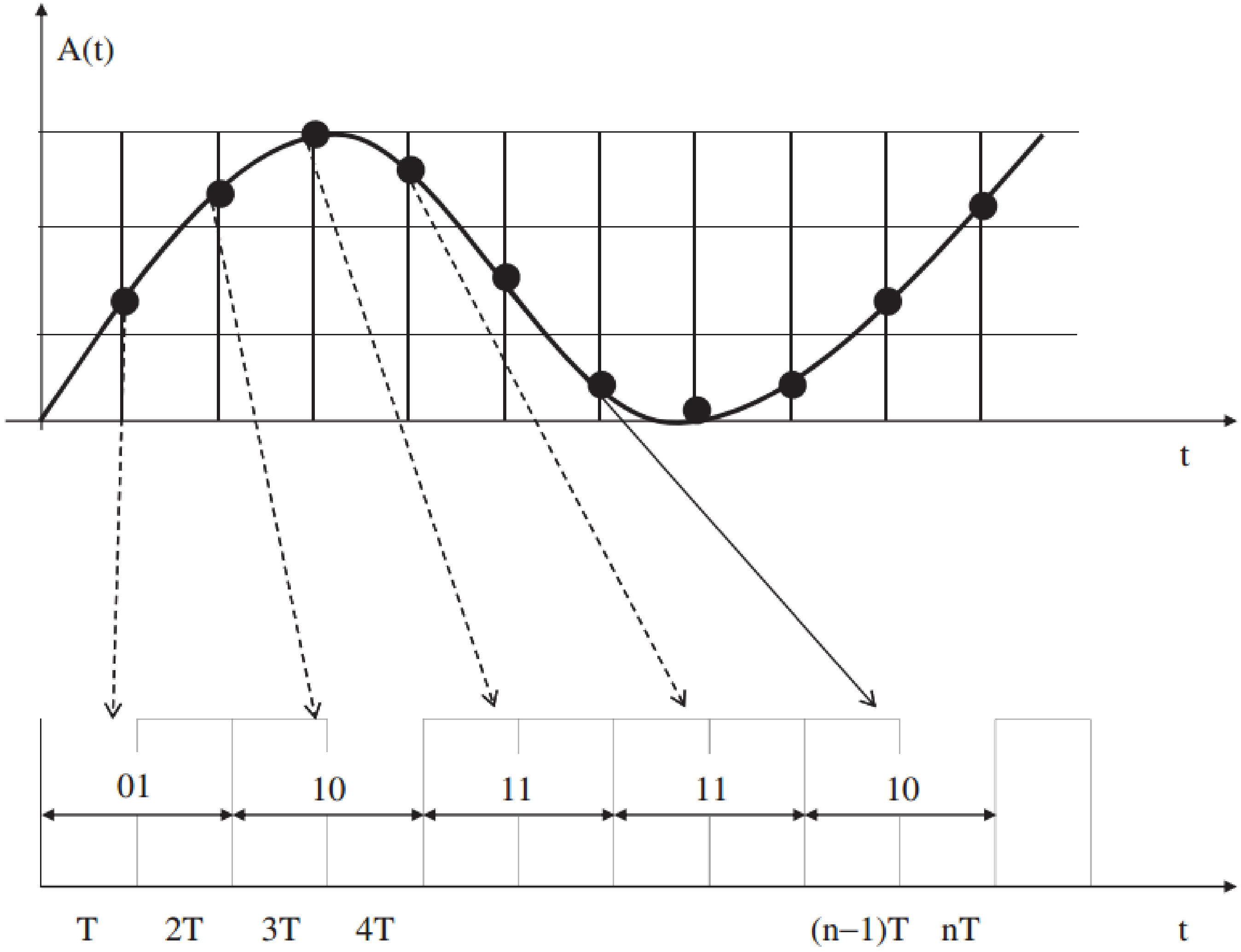

Figure 5.4 A PCM based source encoding.

Το φυσικό επίπεδο

- Delta modulation is a digital pulse modulation technique which has found widespread acceptance in low bit rate digital systems.
- It is a differential encoder and transmits bits of information which describes the difference between successive signal values, as opposed to the actual values of a time-series sequence.
- The difference signal, $V_d(t)$, is produced by first estimating the signal's magnitude based on previous samples ($V_i(t_0)$) and comparing this value with the actual input signal, $V_{in}(t_0)$.
- The polarity of the difference value indicates the polarity of the pulse transmitted.
- The difference signal is a measure of the slope of the signal, which can be achieved by first sampling the analog signal and then by varying the amplitude, width, or the position of the digital signal in accordance with the amplitude of the sampled signal.
- Figure 5.5 illustrates delta modulation.

Το φυσικό επίπεδο

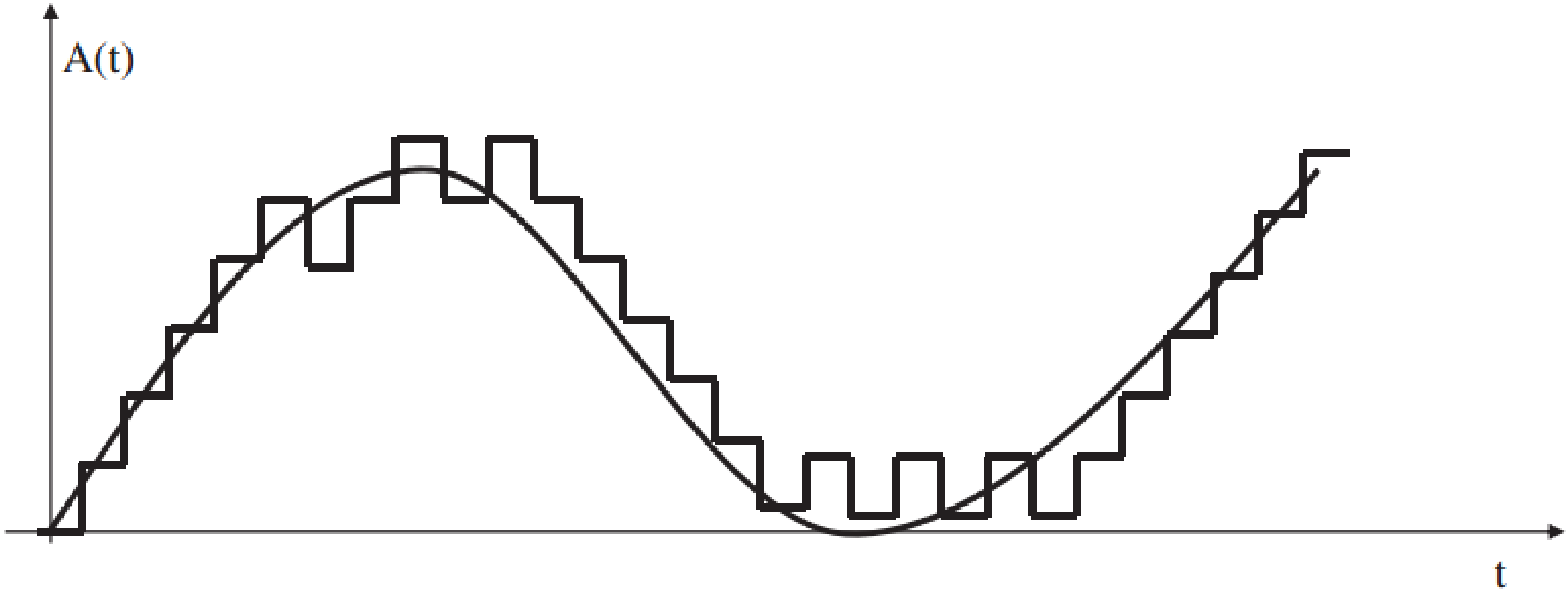

Figure 5.5 Delta encoding.

Το φυσικό επίπεδο

- The main purpose of a channel encoder is to produce a sequence of data that is robust to noise and to provide error detection and forward error correction mechanisms.
- In simple and cheap transceivers, forward error correction is costly and, therefore, the task of channel encoding is limited to the detection of errors in packet transmission.
- The physical channel sets limits to the magnitude and the rate of signal transmission. Figure 5.6 illustrates these restrictions

Το φυσικό επίπεδο

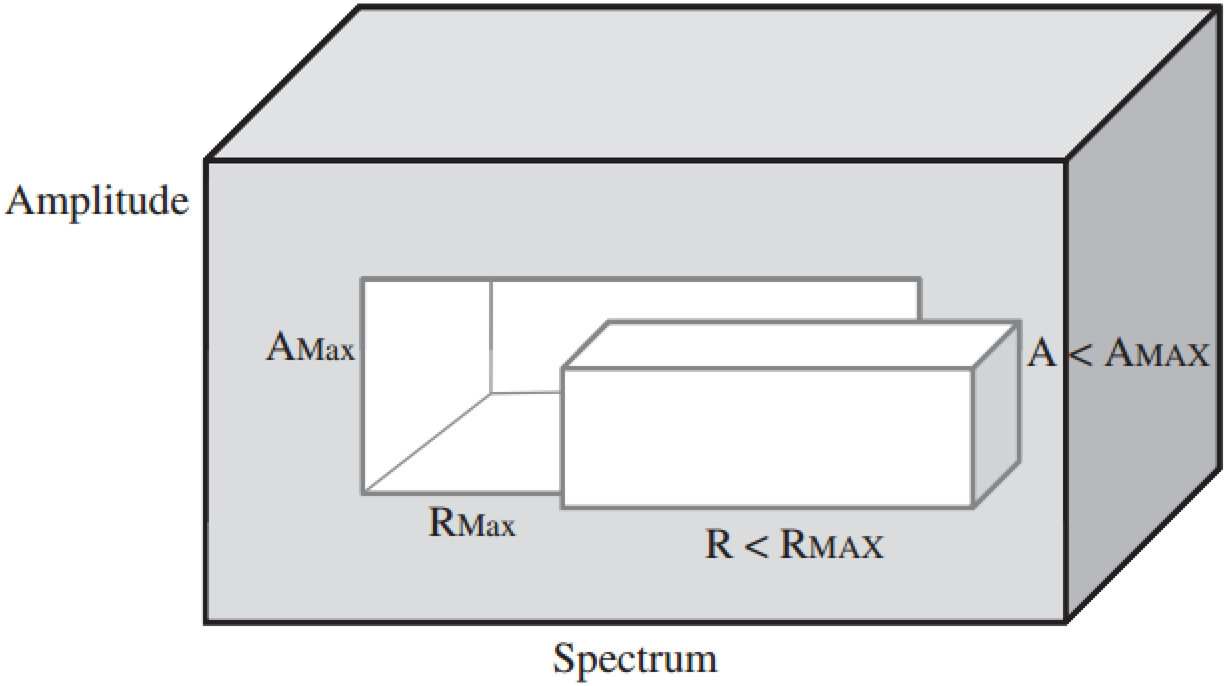

Figure 5.6 Stochastic model of a channel.

Το φυσικό επίπεδο

- According to the Shannon–Hartley theorem, the capacity of a channel to transmit a message without an error is given as:

$$C = B \cdot \log_2\left(1 + \frac{S}{N}\right) \qquad (5.14)$$

C is the channel capacity in bits per second; *B* is the bandwidth of the channel in hertz; *S* is the average signal power over the entire bandwidth, measured in watts; and *N* is the average noise power over the entire bandwidth, measured in watts.

- Equation (5.14) states that for data to be transmitted free of errors, its transmission rate should be below the channel's capacity.
- It also indicates how the signal-to-noise (SNR) ratio, can improve the channel's capacity.

Το φυσικό επίπεδο

- The equation reveals two independent reasons why errors can be introduced during transmission:
  1. Information will be lost if the message is transmitted at a rate higher than the channel’s capacity. This type of error is called *equivocation* in information theory. It is characterized as a subtractive error.
  2. Information will be lost because of noise, which adds irrelevant information into the signal.

Το φυσικό επίπεδο

- A stochastic model of the channel helps to quantify the impact of these two sources of errors.

- Suppose an input sequence of data x_l that can have j distinct values, $x_l \in X = (x_1, x_2, ..., x_j)$, is transmitted through a physical channel.
- Let $P(x_l)$ denote $P(X = x_l)$.
- The channel's output can be decoded with a k-valued alphabet to produce $y_m \in Y = (y_1, y_2, ..., y_k)$.
- Let $P(y_m)$ denote $P(Y = y_m)$.
- At time t_i , the channel generates an output symbol y_i for an input symbol x_i .
- Assuming that the channel distorts the transmitted data, it is possible to model distortion (or transmission probability) as a stochastic process:

Το φυσικό επίπεδο

$$P(y_m|x_l) = P(Y = y_m|X = x_m) \tag{5.15}$$

where, $l = 1, 2, \ldots, j$ and $m = 1, 2, \ldots, k$.

- In the subsequent analysis of the stochastic characteristic of the channel, the following assumptions hold:

- • The channel is discrete, namely, *X* and *Y* have finite sets of symbols.
  • The channel is stationary, namely, *P(ym|xl)* are independent of the time instance, *i*.
  • The channel is memoryless, namely, *P(ym|xl)* are independent of previous inputs and outputs.

Το φυσικό επίπεδο

One way of describing transmission distortion is by using a channel matrix, P_C.

$$P_C = \begin{bmatrix} P(y_1|x_1) & \dots & P(y_k|x_1) \\ \vdots & & \vdots \\ P(y_1|x_j) & \dots & P(y_k|x_j) \end{bmatrix} \tag{5.16}$$

where

$$\sum_{m=1}^{k} p(y_m|x_j) = 1 \,\forall j \tag{5.17}$$

Moreover:

$$P(y_m) = \sum^{j} 1 = 1 P(y_m|x_l).P(x_l) \tag{5.18}$$

or, more generally:

$$(\vec{\mathbf{P}}_y) = (\vec{\mathbf{P}}_x) \cdot [P_C] \tag{5.19}$$

where both $(\vec{\mathbf{P}}_y)$ and $\vec{\mathbf{P}}_x$ are row matrices.

Το φυσικό επίπεδο

A binary symmetric channel (BSC) is a channel model through which bits of information (0 and 1) can be transmitted. The channel transmits a bit of information correctly (regardless of whether 0 or 1 is transmitted) with a probability p and incorrectly (by flipping 1 to 0 and 0 to 1) with a probability $1 - p$. Such a model is displayed in Figure 5.7.

The conditional probabilities for correct and incorrect transmissions are given as:

$$P(y_0|x_0) = P(y_1|x_1) = 1 - p \tag{5.20}$$

$$P(y_1|x_0) = P(y_0|x_1) = p \tag{5.21}$$

The channel matrix of a binary symmetric channel is, therefore given as:

$$P_{\mathrm{BSC}} = \begin{bmatrix} (1-p) & p \\ p & (1-p) \end{bmatrix} \tag{5.22}$$

Το φυσικό επίπεδο

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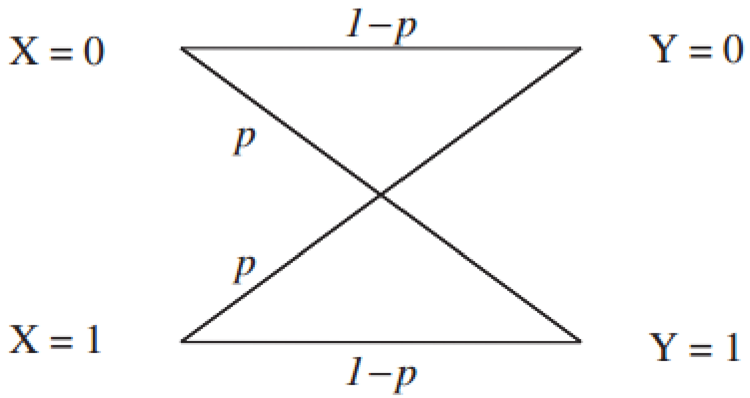

Figure 5.7 A binary symmetric channel model.

Το φυσικό επίπεδο

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Το φυσικό επίπεδο

- In a binary erasure channel (BEC), there is no guarantee that the transmitted bit of information can be received at all (correctly or otherwise).
- Therefore the channel is characterized as a binary input and a ternary output channel.
- The probability of erasure is p and the probability that the information is correctly received is $1 - p$.
- In an erasure channel the probability of error is zero.

Το φυσικό επίπεδο

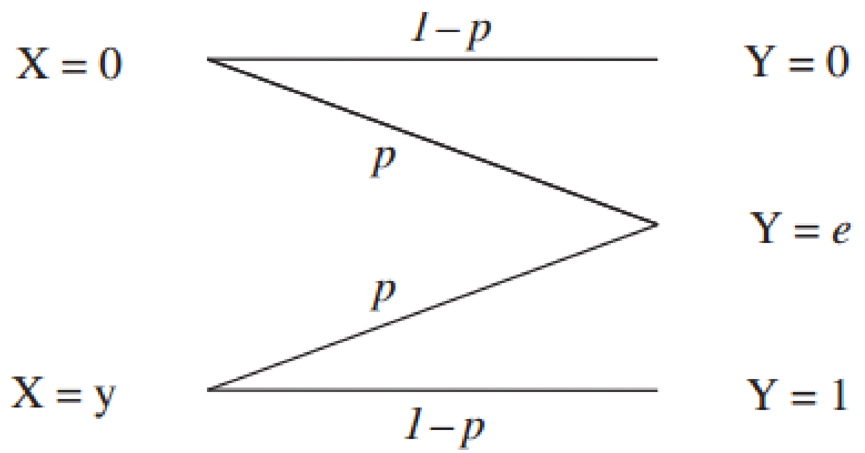

Figure 5.8 A stochastic model of a binary erasure channel.

The channel matrix for a binary erasure channel is given as:

$$P_{\text{BEC}} = \begin{bmatrix} (1-p) & p & 0 \\ 0 & p & (1-p) \end{bmatrix} \tag{5.23}$$

Το φυσικό επίπεδο

- Given the input message, $X : (X, \to Px, H(X))$, the channel matrix, $[PC]$ and the output message, $Y : (Y, Py \to , H(Y))$, it is possible to describe the impact of irrelevance and equivocation as well as the percentage of information that can be transmitted over the channel without an error, which is also called transinformation or mutual information.

- The content of information that can be introduced into the channel due to noise is described as the conditional information content, $I(y|x)$.
- It is the information content of *y* that can be observed provided that *x* is known.
- According to Equation (5.26), a good channel encoder is one that reduces the irrelevance entropy.

Το φυσικό επίπεδο

The conditional entropy is given as:

$$H(y|x) = E_y[I(y|x)] = \sum_{y \in Y} P(y|x) \cdot \log_2 \left(\frac{1}{P(y|x)} \right) \tag{5.24}$$

$P(y|x)$ can be known from the channel matrix $[P_C]$. The average conditional entropy over all input message symbols, $x \in X$, is given by:

$$H(Y|X) = E_x[H(Y|x)] = \sum_{x \in X} P(x) \cdot \sum_{y \in Y} P(y|x) \cdot \log_2 \left(\frac{1}{P(y|x)} \right) \tag{5.25}$$

which is also equal to:

$$H(Y|X) = E_x[H(Y|x)] = \sum_{x \in X} \sum_{y \in Y} P(y|x) \cdot P(x) \cdot \log_2 \left(\frac{1}{P(y|x)} \right) \tag{5.26}$$

From Baye's law, it is clear that:

$$p(x, y) = P(y|x) \cdot P(x) \tag{5.27}$$

Το φυσικό επίπεδο

The content of information that can be lost because of the channel's inherent constraints can be quantified by observing the input x given that the output y is known:

$$H(X|Y) = \sum_{x \in X} \sum_{y \in Y} P(x|y) \cdot P(y) \cdot \log_2 \left(\frac{1}{P(x|y)} \right) \tag{5.28}$$

Once again, applying Baye's conditional probability:

$$P(x|y) = \frac{P(y|x) \cdot P(x)}{P(y)} = \frac{P(y|x) \cdot p(x)}{\sum_{x \in X} P(y|x) \cdot P(x)} \tag{5.29}$$

The conditional probability of Equation (5.29) is also known as the probability of inference or posterior probability. Therefore, equivocation is sometimes called inference entropy. A good channel encoding scheme is one that has a high inference probability. This can be achieved by introducing redundancy during channel encoding.

Το φυσικό επίπεδο

The information content $I(X;Y)$ that overcomes the channel's constraints to reach the destination (the receiver) is called transinformation. Given the input entropy, $H(X)$, and equivocation, $H(X|Y)$, the transinformation is computed as:

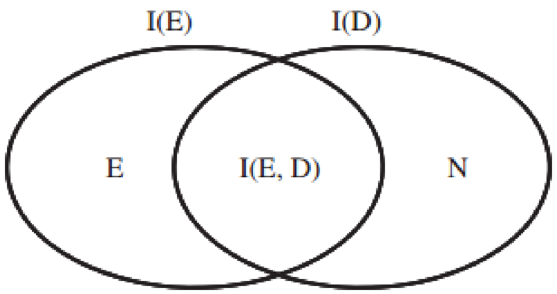

Figure 5.9 Irrelevance, equivocation, and transinformation.

Expanding Equation (5.30) yields:

$$\sum_{x\in X} P(x)\cdot\log_2\left(\frac{1}{P(x)}\right) - \sum_{x\in X}\sum_{y\in Y} P(x,y)\cdot\log_2\left(\frac{1}{P(x|y)}\right) \tag{5.31}$$

Rearranging the terms in Equation (5.31) also yields:

$$H(Y) - H(Y|X) = I(Y;X) \tag{5.32}$$

Irrelevance, equivocation, and transinformation, are summarized in Figure 5.9.

Το φυσικό επίπεδο

- Apart from improving the transinformation of a channel, it is also essential to recognize
  and correct errors during transmission.
- Error recognition can be achieved by permitting the transmitter to transmit only specific types of words.
- If a channel decoder recognizes unknown words, it attempts to correct the error or requests for retransmission (known as automatic repeat request, ARQ).
- In principle, a decoder can correct only *m* number of errors, where *m* depends on the size of the word.
- Error correction, or more precisely, forward error correction, can be achieved by sending *n* bits of information together with *r* control bits.
- The problem with forward error correction is that it slows down transmission.

Το φυσικό επίπεδο

Modulation is a process by which the characteristics (amplitude, frequency, and phase) of a carrier signal are modified according to the message (a baseband) signal.

Modulation has several advantages:

- the message signal will become resilient to noise;
- the channel’s spectrum can be used efficiently; and
- signal detection will be simple.

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