MAXIMUM DATA RATE DETERMINATION OF A TELEPHONE TRANSMISSION CHANNEL

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ABSTRACT

The transmission channel is an electrical medium that bridges the distance from the source to the destination. It may be a pair of wires, a coaxial cable, a radio path, or an optical fiber. Every channel introduces some amount of transmission loss or attenuation and, therefore, the transmitted power progressively decreases with increasing distance. The signal is also distorted in the transmission channel because of different attenuation at different frequencies. Signals usually contain components at many frequencies and if some are attenuated and some are not, the shape of the signal changes. This change is known as distortion. Note that a transmission channel often includes many speech or data channels that are multiplexed into the same cable pair or fiber. In this paper we present the determination of maximum data rate of a telephone channel.

Keywords: Bandwidth, Data Rate, Symbol Rate, Transmission Channel

1. INTRODUCTION

Voice communication usually requires a constant data rate of 64 Kbps or less and high-resolution video a constant data rate of 2 Mbps or higher over the network. Characteristics of data communication are very different, for example, file transfer requires high-bit-rate transmission only during download, and high resolution graphics on a Web page require high-data-rate transmission only when we download a new page. When we are reading a Web page we do not need transmission capacity at all. To define data transmission capacity, we sometimes use the term *bandwidth* instead of *data rate* because these terms are closely related to each other.

Various unwanted factors impact the transmission of a signal. Attenuation is undesirable because it reduces signal strength at the receiver. Even more serious problems are distortion, interference, and noise, the last of which appears as alterations of the signal shape. To decrease the influence of noise, the receiver always includes a filter that passes through only the frequency band of message frequencies and disables the propagation of out-of-band

A fundamental limit exists for the data rate through any transmission channels are mainly restricting factors that are the bandwidth and the noise of the channel.

2. SYMBOL RATE (BAUD RATE) AND BANDWIDTH

Communication requires a sufficient transmission bandwidth to accommodate the signal spectrum; otherwise, severe distortion will result. For example, a bandwidth of several megahertz is needed for an analog television video signal, whereas the much slower variations of a telephone speech signal fit into a 4-kHz frequency band. The voice signal, which is the most common message in telecommunications network, does not look similar to a pure cosine wave. It contains many cosine waves with different frequencies, amplitudes, and phases combined together. The range of frequencies that is needed for a good enough quality of voice, so that the speaker can be recognized, was defined to be the range from 300 to 3,400 Hz. This means that the bandwidth of the telephone channel through the network is 3,400 – 300 Hz = 3.1 kHz, as shown in Figure 1.

A human voice contains much higher frequencies, but this bandwidth was defined as a compromise between quality and cost. It is wide enough to recognize the speaker, which was one requirement for telephone channel. Bandwidth is not strictly limited in practice, but signal attenuation increases heavily at the lower and upper cutoff frequencies. For speech, channel cutoff frequencies are 300 and 3.4 Hz, as shown in Figure 1.

The bandwidth is normally measured from the points where the signal power drops to half from its maximum power. Attenuation or loss of channel is given as a logarithmic measure called a decibel (dB), and half power points correspond to a 3-dB loss. Bandwidth, together with noise, is the major factor that determines the information-carrying capacity of a telecommunications channel. The term *bandwidth* is often used instead of *data rate* because they are closely related.

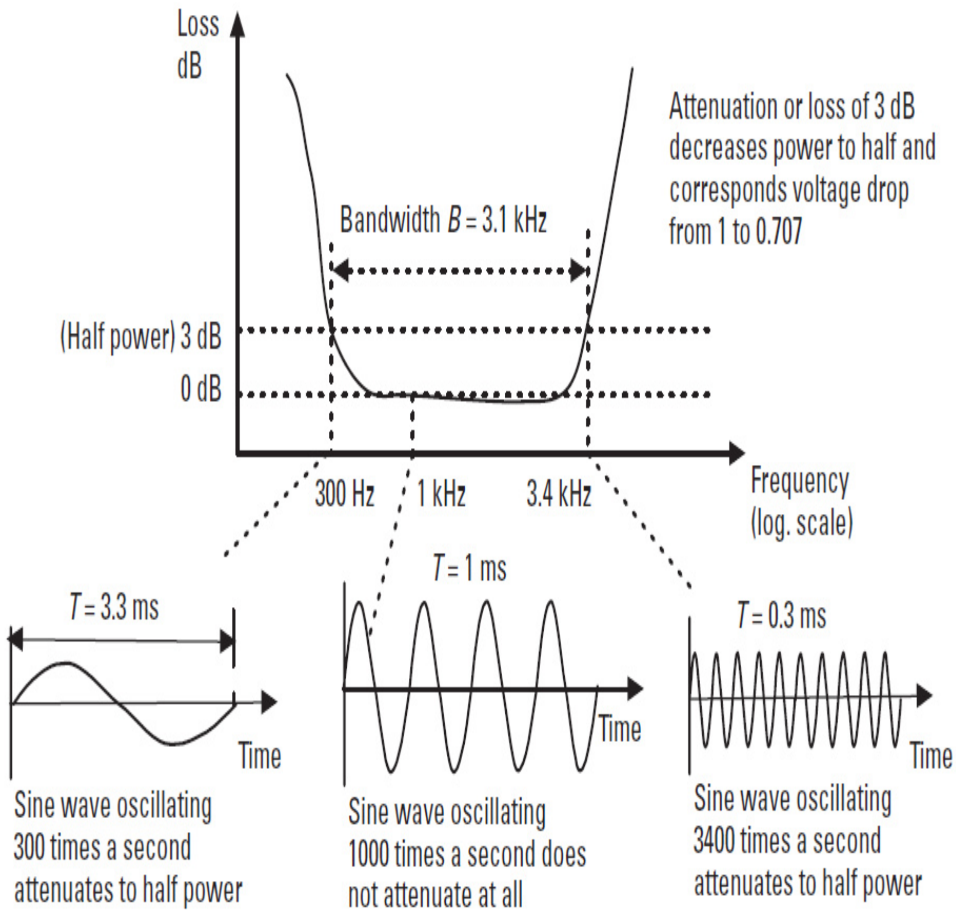

Figure 1: Bandwidth of the telephone speech channel.

Every communication channel has a finite bandwidth. The higher the data rate to be transmitted, the shorter the digital pulses that can be used, as we know from previous experience.

The shorter the pulses used for transmission, the wider the bandwidth required, as we saw in Figure 2. When a signal changes rapidly in time, its frequency content or spectrum extends over a wide frequency range and we say that the signal has a wide bandwidth.

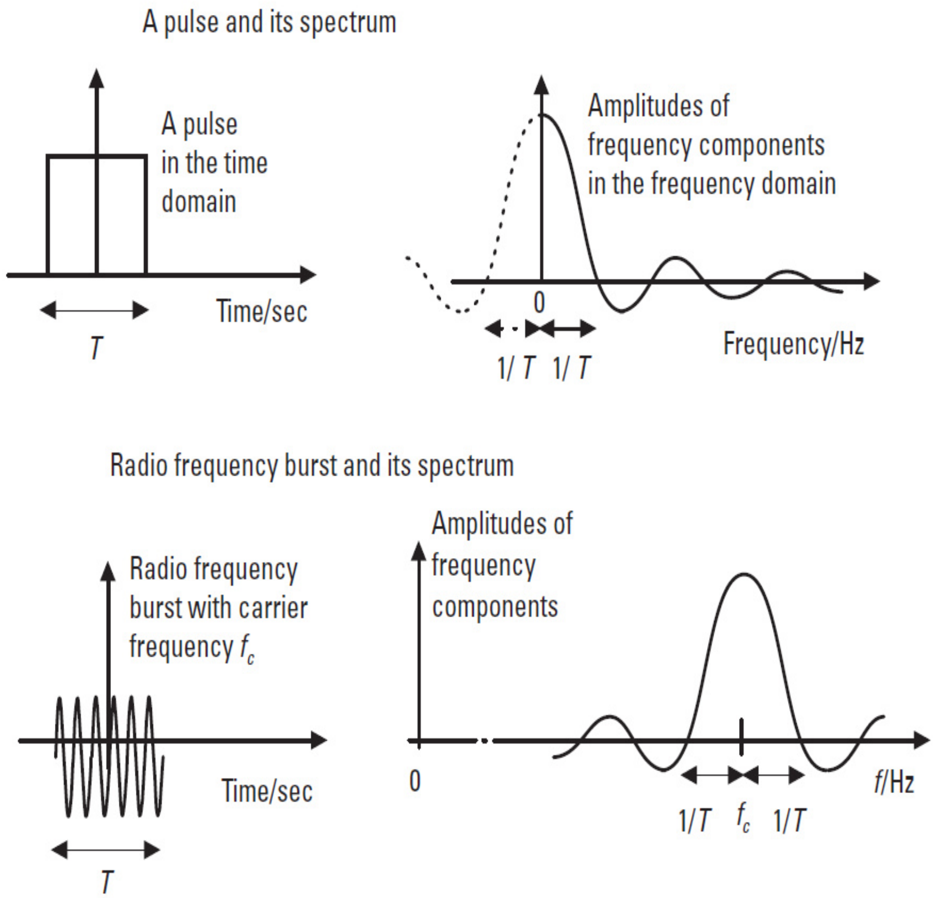

Figure 2: Signals in the time domain and the spectrum.

Figure 3 shows the shape of a rectangular pulse with duration T before and after it passed through an ideal lowpass channel of bandwidth B. For example, if the duration of the pulse $T = 1$ ms, distorted pulses are shown in the figure for the channel with bandwidths $B = 2 \cdot 1/T = 2$ kHz, $B = 1/T = 1$ kHz, $B = 1/2 \cdot 1/T = 500$ Hz, and $B = 250$ Hz. If the next pulse is sent immediately after the one in the figure, the detection of the pulse value will be impossible if the bandwidth is too narrow. The spread of pulses over other pulses, which disturbs detection of other pulses in the sequence, is called *intersymbol interference*.

In baseband transmission, a digital signal with r symbols per second, bauds, requires the transmission bandwidth B to be in hertz:

$B \geq r/2$ …………… (1)

Thus the available bandwidth in hertz determines the maximum symbolrate in bauds. Note that the symbol is not necessarily the same as the bit,but it can carry a set of bits if it is allowed to get many different values.We can find the theoretical maximum of the symbol or baud rate withthe help of a special pulse called the *sinc pulse*. The shape of the sinc pulse isdrawn in Figure 3 and it has zero crossings at regular intervals $1/(2W)$.With the help of Fourier analysis, we can show that this kind of pulse has nospectral components at frequencies higher than W. If the channel is an ideal lowpass channel with a bandwidth higher than W, it is suitable for transmittingsinc pulses that have their first zero crossing at $t = 1/2W$ without distortion.The shape of the pulse remains the same because all frequencycomponents are the same at the output as at the input of the channel.

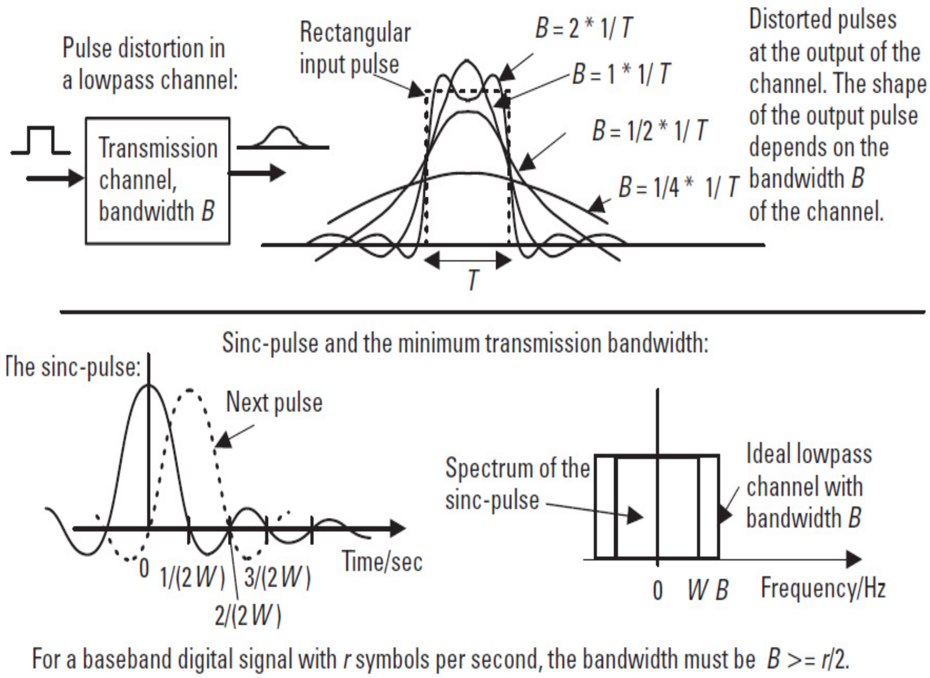

Figure 3: Symbol rate (baud rate) and bandwidth.

The sinc pulses have zero crossings at regular periods in time. Theseperiods are $1/(2W)$ seconds for a sinc pulse with a spectrum up to frequency$W$ as shown in Figure 3. We can transmit the next pulse at the timeinstant $1/(2W)$ so that the previous pulse has no influence on the receptionbecause it crosses zero at that time instant. The decision for the value of thepulse is made in the receiver exactly at time instants $n \cdot 1/(2W)$, where $n = \pm 1, \pm 2, \pm 3, \ldots$.

The time between pulses $T = 1/(2W)$, which makes datarate $r = 1/T = 2W$. If we now increase the data rate so that $W \rightarrow B$, the timebetween pulses becomes $T \rightarrow 1/(2B)$; $r \rightarrow 1/T = 2B$, which gives the theoreticalmaximum rate for transmission of symbols and we can say that the symbolrate and bandwidth are related according to $r \leq 2B$ or $B \geq r/2$.

This kind of pulse does not exist in reality, but the result gives the theoreticalmaximum symbol rate, which we can never exceed, through a lowpasschannel. In real-life systems quite similar pulse shapes are in use and typicallya 1.5 to 2 times wider bandwidth is needed.

3. SYMBOL RATE AND BIT RATE

In digital communications a set of discrete symbols is employed. Binary systemshave only two values represented by binary digits 1 and 0. In the previoussection we found that the fundamental limit of the symbol rate is twicethe bandwidth of the channel. With the help of the symbols with multiplevalues the data rate, in bits per second, can be increased. As an example, withfour pulse values we could transmit the equivalent of 2-bit binary words 00,01, 10, and 11.

Thus each pulse would carry the information of 2 bits andone symbol per second (1 baud) would correspond to 2 bps.If we use a sinc pulse as in Figure 3, the preceding and followingpulses do not influence the detection of a transmitted pulse, because eachreceived pulse is measured at a zero crossing point $n \cdot 1/(2W)$ of the otherpulses. We may increase the number of peak values of sinc pulses from two tofour, from four to eight, for example, in order to increase the bit rate whilekeeping the symbol rate unchanged. Figure 4 shows a simple examplewhere symbols are rectangular pulses with four symbol values and each symbolcarries two bits ($k = 2$) of information. Generally, the bit rate depends onmodulation rate according to

$r_b = k \cdot r$ bps ………………..(2)

Where$k$ represents the number of bits encoded into each symbol. Then thenumber of symbol values is $M = 2^k$and the bit rate is given as $r_b = r \log_2 M$bps.

In the example of Figure 4, the number of symbol values is $M = 2k = 2^2 = 4$, and the bit rate $r_b = k \cdot r$ bps= $2r$ bps. Then the symbol rate of 1 Kbaud makes the bit rate 2 Kbps.

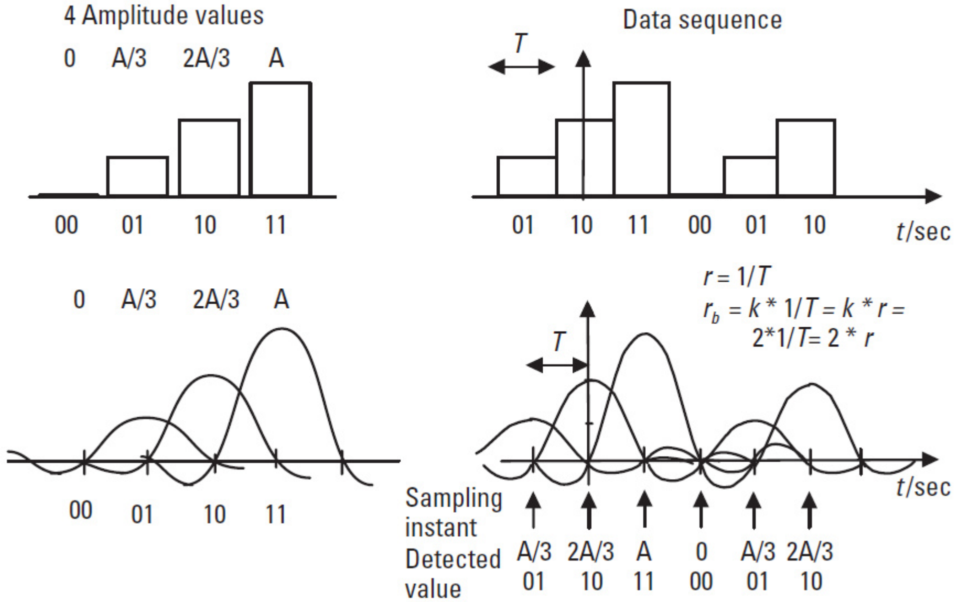

Figure 4: Symbol rate and bit rate.

The unit of symbol rate, sometimes called the modulation rate, isbauds (symbols per second). Note that the transmission rate in bauds mayrepresent a much higher transmission rate in bits per second. Table 4.1shows how the bit rate of a system depends on the number of symbol values.Figure 4 also shows a data sequence of sinc pulses with four amplitudevalues. The required bandwidth for pulses in this sequence is the minimumbandwidth B= $r/2 = 1/(2T)$ according to (1) and Figure 3. Whenpulses are detected by sampling as shown in Figure 4 each pulse can bedetected independently because values of all other pulses are equal to zero.In the preceding examples, the amplitudes of the pulses contain theinformation. This is the principle of PAM. This is notthe only alternative. We can use other characteristics of the signal as well tocreate multiple symbol values, for example, the phase of a carrier, as we did inthe case of QPSK and 8-PSK in Figure 7. There we used a certainmodulation rate ***r***in bauds (how many times the phase can change in a second),which defines a required bandwidth. For QPSK 2 bits ($k = 2$) are encoded into each symbol and the bit rate is two times the modulation rate.

For 8-PSK, $k = 3$ and $r_b = 3r$. The 16-QAM example in Figure 6 used 16combinations of carrier amplitude and phase amplitude values and the bit rate is four times the modulation rate. As we can see from Table 4.1, by increasing the number of differentsymbols used in the system the data rate could be increased without a limit ifthere were no other limitations than bandwidth. This is not possible in practicebecause of the noise.

Table 4.1Bit Rate of a System Using Multiple Symbol Values

No of Bits, k, enclosed in each symbol	Number of symbol values, M	Bit rate compared with symbol rate
1	**2**	**Same as symbol rate**
2	**4**	**2 X symbol rate**
3	**8**	**3 X symbol rate**
4	**16**	**4 X symbol rate**
5	**32**	**5 X symbol rate**
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8	**256**	**8 X symbol rate**
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4. MAXIMUM CAPACITY OF A TRANSMISSION CHANNEL

We saw previously that the bandwidth of a channel sets the limit to the symbolrate in bauds but not to the information data rate. In 1948, Claude Shannonpublished a study of the theoretical maximum data rate in the case of achannel subject to random (thermal) noise.

We measure a noise relative to a signal in terms of the S/N. Noisedegrades fidelity in analog communication and produces errors in digitalcommunication. The S/N is usually expressed in decibels as

$S/N_{dB} = 10 \log_{10}(S/N)$ dB ……………….(3)

Taking both bandwidth and noise into account, Shannon stated thatthe error-free bit rate through any transmission channel cannot exceed themaximum capacity *C* of the channel given by:

$C = B \log_2(1+ S/N)$ ………………….(4)

Where*C* is the maximum information data rate in bits per second;

B is thebandwidth in hertz;
S is the signal power;
N is the noise power,and
S/*N is* theS/N power ratio (absolute power ratio, not in decibels).

Equation (4) gives a theoretical limit for the data rate with an arbitrarilylow error rate when an arbitrarily long error correction code is used. Italso assumes that the signal has a Gaussian distribution as does the noise,which is not the case in practice. The influence of bandwidth and noise inthe case of binary and multiple value signaling is summarized in Figure 5.

The signal power and, thus, the highest value of the signal are alwaysrestricted to a certain maximum value. Then, the more symbol values we use,the closer they are to each other, and the lower noise level can cause errors.Thus, a higher bit rate requires a wider bandwidth that allows a highersymbol rate. Alternatively, a better S/N is required to allow for more symbolvalues.

The example in Figure 5 shows what happens to the distancebetween symbol values when the maximum amplitude is *A*and four symbolvalues are used instead of binary symbols that have only two values. In ourexamples we have used symbols with different amplitudes. This transmissionscheme is called PAM, as discussed earlier.

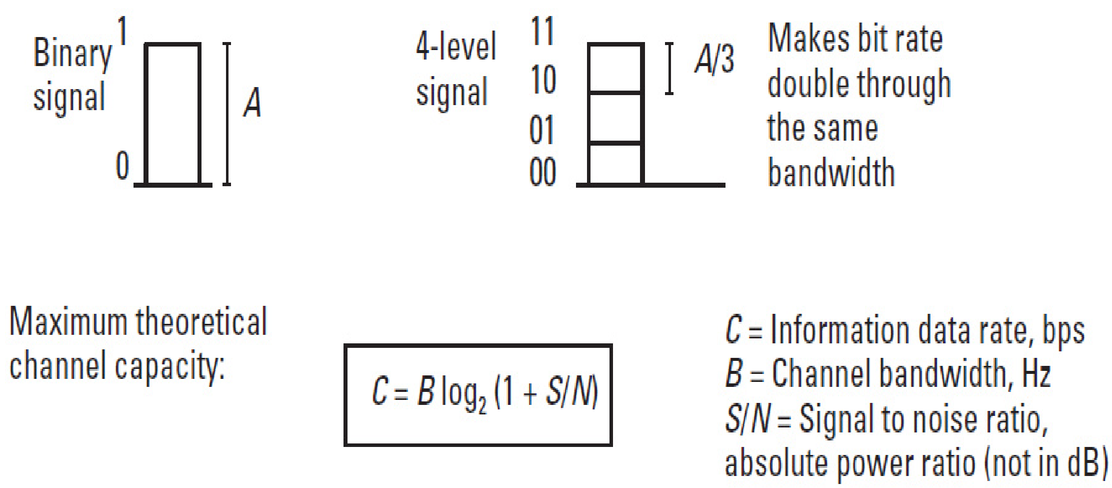

Figure 5: The maximum capacity of a transmission channel.

Transmission of this type ofpulses without CW modulation is called baseband transmission.In radio systems or modems that use CW modulation, different phasesof a carrier wave represent different symbol values. In the Figure 6digital phase modulation methods, BPSK, QPSK, and 8-PSK all require thesame bandwidth if symbol rate is the same, but the information data rate forQPSK is double and for 8-PSK triple compared with BPSK. The cost we have to pay is reduced noise tolerance because signals become closer to eachother as more symbol values or different signals are used. It is not usually reasonableto use more than eight phases; instead, we use different amplitudes asin 16-QAM in Figure 6.

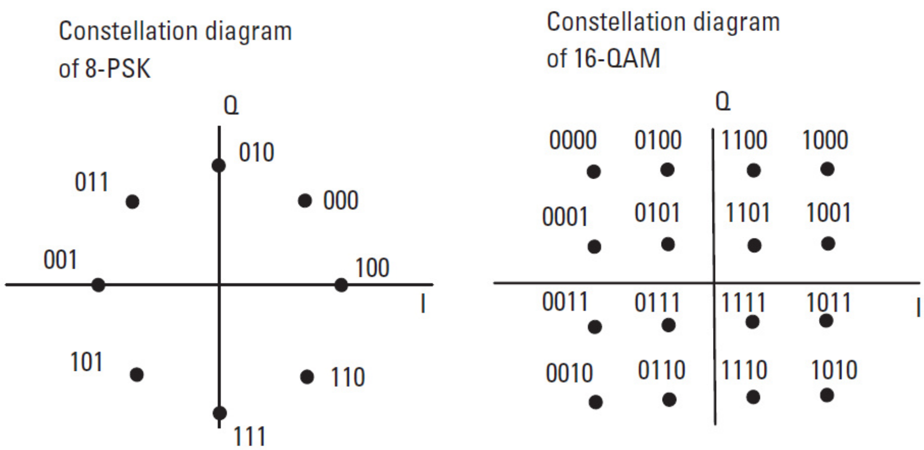

Figure .6 8-PSK and 16-QAM.

The 16-QAM tolerates more noise than 16-PSKbecause with the same average signal power distances between signals can beincreased. However, if we would analyze noise tolerance in more detail, wecould form a general rule stating that the increase in the number of signals inuse reduces noise tolerance.

In low-noise channels, such as telephone voicechannels, many different signals can be used but in high-interference channels,such as radio channels for cellular systems, binary symbols are often thebest choice. However, modulation moves the spectrum of the pulse from low frequenciesto carrier frequencies, and the bandwidth is typically doubled whencompared with baseband systems as was shown in Figure 7 below.

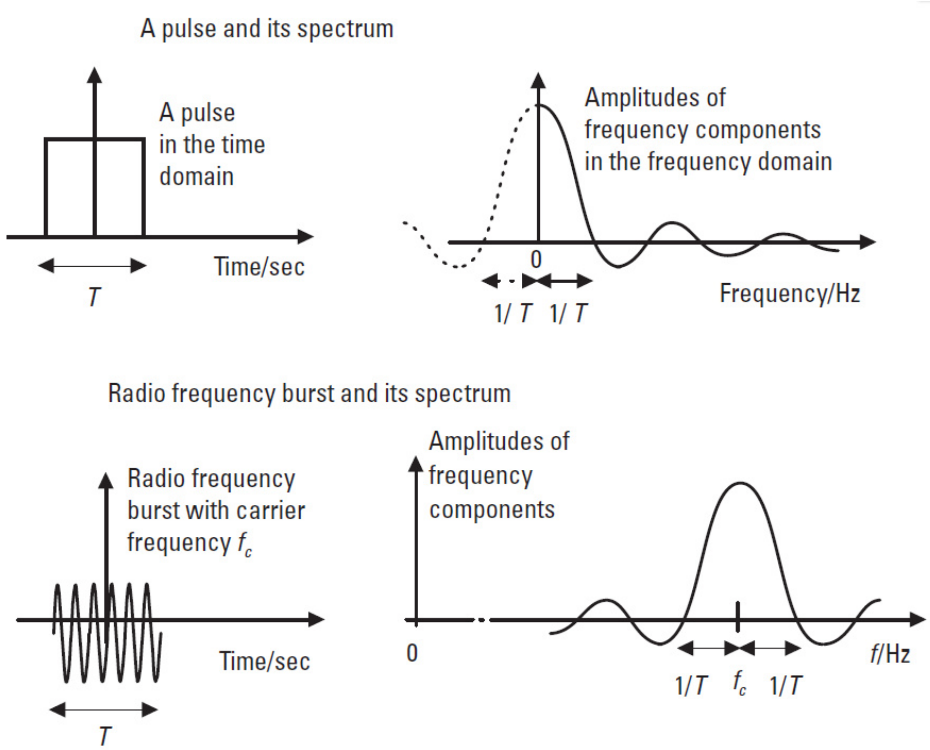

Figure 7: Spectrum of the pulse from low frequencies to carrier frequencies

This is why thesymbol rate in radio systems is less than or equal to the transmission bandwidth,that is:

$r \leq BT$ ………… (5)

Where$r$ is the symbol rate in bauds and BT is the transmission bandwidth inhertz.The accurate requirement of bandwidth depends on the modulationscheme in use.

5. CONCLUSION

In this paper we presented the determination of maximum data rate of a telephone channel with no losses or with no effect of Noice. If all other papameters are tacken in to account, there will be a change in the data rate, which is consider for future work.

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