Bits, Symbols, Bauds, and Bandwidth

Roger L. Freeman

ABSTRACT The use of clear, concise, and unambiguous language in telecommunications engineering is vital to communicate a desired meaning and understanding. Terminology should be based on rigorous engineering principles and traceable to well known and dependable sources. This article deals with four everyday terms commonly encountered in the popular semi-technical press as well as in serious engineering periodicals. The misuse of these and other terms can lead to low-balled cost proposals, court litigation, and patent infringement cases. Imagine how an engineering student can be confused when she/he typically encounters bandwidth measured in bits per second.

When I started in the field of data communications, I always wondered *how many bits per second of data could fit into a unit of bandwidth*. The question still seems to bother a lot of people, as my seminar experience bears out. In the process of relating bit rates in *bits per second* to *bandwidth* measured in hertz, we will encounter two other terms that are associated with bandwidth and bit rate. These are *symbols per second* and *baud*. The misuse of these four terms is very common. I am sure you have heard someone say that he/she is using a 33.6 kilobaud modem for operation on a nominal 4 kHz voice channel? This is the blatant misuse of a perfectly good term: baud. Another one is "I'm going to increase the bandwidth to 384 kb/s." This is followed by "bandwidth on demand." We come equipped with a perfectly good language of telecommunications. Admittedly, there remain some ambiguities that we probably cannot overcome. However, there are others that have grown with common use. The purpose of this article is to resolve the ambiguities and poor usage of four common terms: bandwidth, bit rate, symbol, and baud.

Thus, we will first define bandwidth from several perspectives. Then there will be a very brief review of two well-known theoreticians' views on the bit rate/bandwidth question. Finally, we will delve into some practical matters and attempt to better define bits, symbols, bauds, and bandwidth.

BANDWIDTH DEFINED

Bandwidth plays a vital role in telecommunications. Its unit of measure is the hertz. It can be defined as the range between the lowest and highest frequencies used for a particular application. One of the most common definitions deals with signal power. For filters, attenuators, amplifiers, transmission lines, and other electromagnetic active and passive equipment, these limits are generally taken where a signal will fall 3 dB below the average (power) level in the passband, or below the level at a reference frequency [1]. The voice channel as used on a subscriber loop is one notable exception. In North America it is specifically defined at the 10-dB points about the reference frequency of 1000 Hz, approximately in the band 200 to 3200 Hz [2, 3]. Anywhere else in the network, the bandwidth definition follows the 3 dB rule [4]. The International Consultative Committee for Telephone and Telegraph (CCITT) voice channel occupies the band 300 to 3400 Hz. To further confuse matters, we call this the nominal 4-kHz voice channel. The bandwidth of the CCITT voice channel, of course, is 3100 Hz (not bits per second). It is based on the 3-dB power rule.

The 3-dB bandwidth concept is shown in Fig. 1.

As Frank Amoroso states in [5], "No single definition suffices." Thus, there are at least four other ways we can define bandwidth [5, 6].

Radio regulating agencies, as one might imagine, have a very strict definition, which I like to call the 20 dB rule. Others call it the 99 percent power containment rule. For example, the U.S. Federal Communications Commission has adopted it. Simply stated, the occupied bandwidth (of a radio emission) is the band which leaves exactly 0.5 percent of the signal power above the upper band limit and exactly 0.5 percent of the signal power below the band limit. Thus, 99 percent of the signal power is inside the occupied band [5].

Still another definition is the *bounded power spectral density* [5]. This states that everywhere outside of s specified band, the power must have fallen at least to a certain stated level below that found in the band center. Typical attenuation levels might be 35 or 50 dB. There are also the "null-to-null bandwidth" and "noise bandwidth" definitions. Bandwidth can be defined in any number of ways depending on the application; thus, we do not believe our listing is exhaustive.

THE BIT AND BITS PER SECOND

The bit is an abbreviation of *binary digit*. Some call it the lowest unit of information. We will all remember that it can take on one of two states, commonly called 1 or 0. Just a 1 or a 0 is not really rich with information. It is, however, sufficient for some needs such as supervisory signaling. For other data users, we turn to source codes such as ITA#2 (CCITT), American Standard Code for Information Interchange (ASCII) sponsored by the American National Standards Institute (ANSI), and IBM's 8-bit EBCDIC (extended binary coded decimal interchange code). In any case, such codes develop characters from consecutive, contiguous bits, 5, 6, 7, or 8 in a row. In many circumstances 8-bit codes are preferred because 8 is a binary-related number, and fits well with digital processors [7].

In any case, we measure the information transmission capacity in the binary domain by *bits per second*. We assume that these bits all have the same bit period and are contiguous. This is only true in synchronous transmission, where one bit is directly followed by the next throughout a frame [7].

Relating Bit Rate to Bandwidth

First and Second Bandwidth Approximations

If we remain in the binary domain (i.e., we deal only with two states, 1s and 0s), we will often approximate bandwidth as 1 b/Hz of bandwidth. This first approximation roughly allows for nominal impairments such as amplitude-frequency response, phase distortion, and noise. It is a *first* approximation.

For example, 2400 b/s will require a bandwidth of 2400 Hz. Another example is DS1, which with this first approximation will require 1.544 MHz to support 1.544 Mb/s. This tells us that the voice channel, which we said has a bandwidth of 3100 Hz, can support no more than 3100 b/s. How, pray tell, can we transmit 33,600 b/s in such a small bandwidth? Certainly not with our 1 b/Hz approximation. We will get into that magic a little later. First let us look at what two of the great pioneers in information theory have done to provide a second approximation.

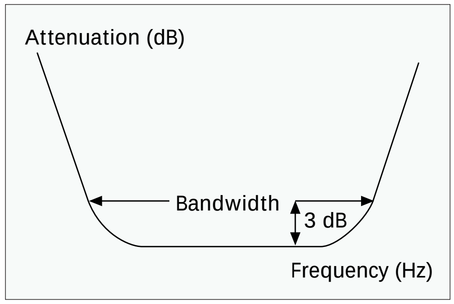


Figure 1. *The concept of 3 dB power bandwidth.*

H. Nyquist of Bell Telephone Laboratories [8, 9] was a trailblazer in information theory. He essentially stated that theoretical error-free transmission of two bits of information can be transported in 1 Hz of bandwidth. This is often called the *Nyquist rate* or *Nyquist bandwidth*. In other words, if we have 1000 Hz of bandwidth, theoretically it can transport 2000 b/s "error-free." In stating this, Nyquist's only concern was intersymbol interference. His model was a low-pass filter cutting off at f_1 Hz. Transmitting bits without intersymbol interference is possible at a rate of $2f_1$ independent values per second through such an ideal low-pass filter cutting off at f_1 Hz.

Claude Shannon, also of Bell Telephone Laboratories, came up with a different model [10]. Shannon's traditional formula relates channel capacity C (binary digits or bits per second) to bandwidth W (in hertz) by the following:

$$C = W\mathrm{Log}_2[(P_{\mathrm{max}} - P_{\mathrm{noise}})/P_{\mathrm{noise}}] \text{ b/s}$$

where P_{max} is the signal power and P_{noise}, in this case, is thermal noise power.

This formula is also written as

$$C = W\mathrm{Log}_2(1 + \mathrm{S/N})$$

Signal-to-noise ratio (SNR) in this case must be expressed as a numeric. Suppose SNR was 30 dB. Its numeric is 1000. Let the bandwidth be 3000 Hz; then

$$C = 3000 \times \mathrm{Log}_2(1 + 10^3) \text{ or (approximately)}$$
$$= 30{,}000 \text{ b/s}$$

Often we will hear of the *Shannon limit* [7] in reference to the above relationship. As mentioned previously, one of the principal impairments in the transmission channel is phase distortion. It is mainly this impairment that restricts us from reaching the Shannon limit. In a bandwidth-limited channel, as the bit rate increases beyond a certain point, intersymbol interference begins to impact bit error performance [11].

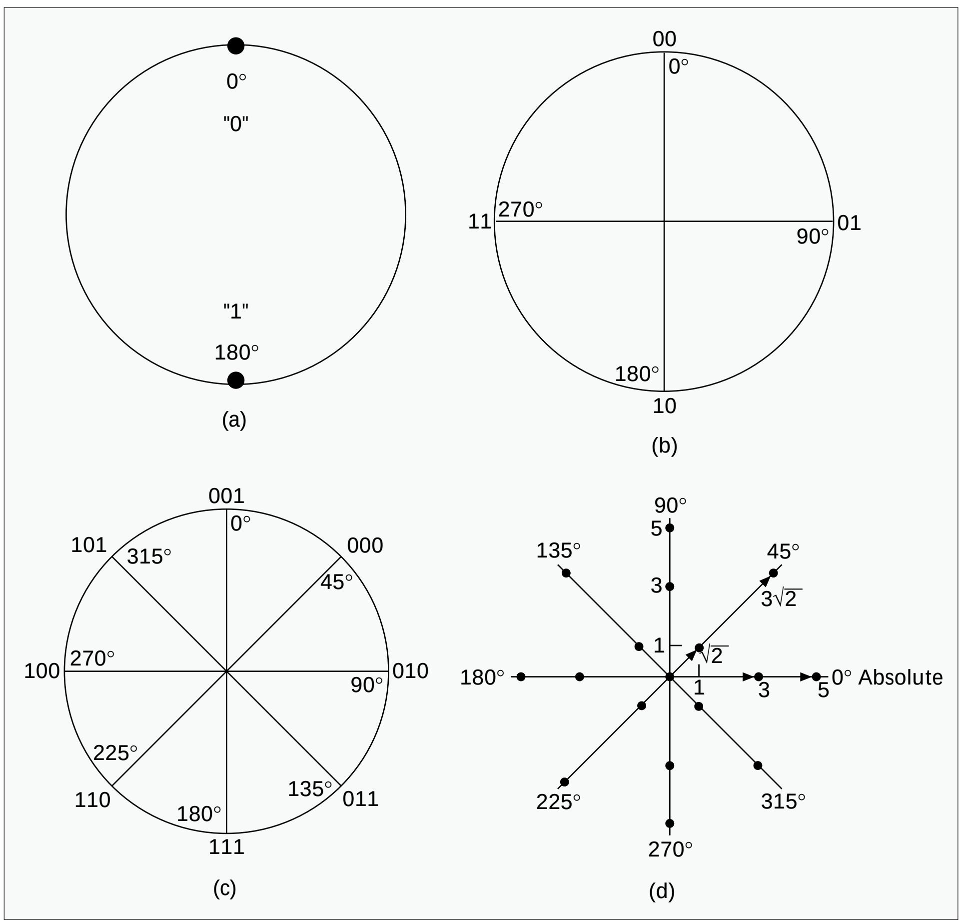
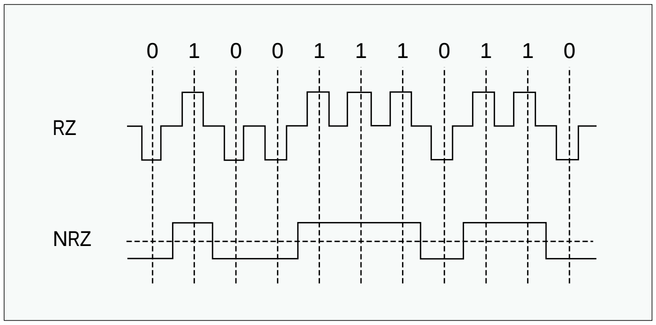


Figure 2. *a) A spatial representation of a data tone with BPSK modulation — only 1 bit is transmitted per transition; b) a spatial representation of a data tone with QPSK modulation — 2 bits are transmitted for every transition; c) a spatial representation of 8-ary PSK — 3 bits are transmitted for every transition; d) a spatial representation of a data tone using a hybrid modulation scheme called QAM transmitting at 9600 b/s based on CCITT Rec. V.29 — for each transition 4 bits are transmitted.*

Squeezing More Bits per Hz of Bandwidth

Bit Packing

Bit packing allows us to transmit at greater bit rates than the 1 b/Hz implies or even the 2 b/Hz suggested by Nyquist. The data rate a transmission medium can support is determined by the number of transitions per second that are placed on the medium and certain of its physical characteristics. To increase the data rate measured in b/s, we



■ **Figure 3.** *RZ and NRZ binary baseband waveforms compared. In the figure there are 22 transitions for the RZ waveform compared to only six transitions for the NRZ waveform based on the same binary sequence. The "Z" in RZ and NRZ refers to the zero voltage line.*

can turn to higher-level modulation techniques such as quadrature phase-shift keying (QPSK). With binary phase shift keying (BPSK), one bit is transmitted for each transition (i.e., change of state) (Fig. 2a). QPSK allows us to theoretically transmit two bits for each transition. Figure 2b illustrates this concept. Nearly all 2400 b/s modems use QPSK. The term *baud* is the measure of modulation rate or the number of transitions per second. In the case of the 2400 b/s modem, the modulation rate is 1200 baud. For every transition with QPSK, two bits are transmitted.

8-ary PSK carries this concept one step further. Three bits are transmitted for each transition. This idea is illustrated in Fig. 2c. The 4800 b/s modem specified in CCITT Recommendation V.27 specifies 8-PSK modulation. The modulation rate of this modem is 1600 baud.

Another example is the 9600 b/s modem specified in CCITT Recommendation V.29. Its modulation rate is 2400 baud. It achieves 4 b/Hz bit packing by using a hybrid combination of 8-ary PSK and two amplitude levels [7]. This is illustrated in Fig. 2d.

There are no free lunches, as I have been told. Thus, there is a price for turning to *M*-ary modulation techniques. This price, following the Shannon capacity formula, is paid in increased SNR as *M* increases for a given error performance. There is one exception, however. No increase in signal power is necessary as we move from BPSK to QPSK for a given error performance. This is because a QPSK modulator is made up of two BPSK modulators, one being 90 degrees out of phase with the other.

Bauds and bits per second are synonymous only in the binary domain. In the *M*-ary domain they are not synonymous. As you are probably aware, the term *baud* derives from a Frenchman who is considered the father of automatic telegraphy. His name was Emile Baudot.

Even in the binary domain, care must be exercised in estimating bandwidth. This is particularly true with nominal binary baseband waveforms such as the return-to-zero (RZ) waveform which has many more transitions per second than the familiar non-return-to-zero (NRZ) waveform. These two binary waveforms are shown in Fig. 3. Bandwidth is related to transitions per second, not bits per second. Also, useful bandwidth for digital transmission is related to two impairments: group delay and amplitude distortion. We cannot increase the bandwidth of a given transmission medium. We can, however, mitigate the effects of these impairments with amplitude and delay equalizers to take better advantage of a transmission medium's bandwidth [7].

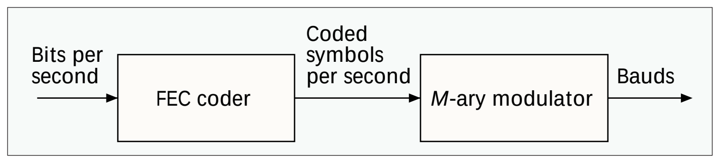
Currently there are three specialty areas in telecommunications where we would like to pack as many bits as possible per unit of bandwidth.

The Voice Channel — looking into the telecommunications network where only 3100 Hz bandwidth is available. The V.34 bis modem can provide a 33.6 kb/s bit rate. If the channel terminates in a pulse code modulated (PCM) channel bank, the U-2 modem has the capability of 56 kb/s.

The Subscriber Loop — looking toward the subscriber (called *downstream*). The pressure to increase bit rate here derives from Internet service. Various methods are now available to increase the data rate from 160/192 kb/s of ISDN to 6 Mb/s or more of the various digital subscriber loop (xDSL) schemes. Loop length and its condition are critical constraints.

Digital Line-of-Sight Microwave — The RF spectrum is a shared resource. Emission sources wasteful of bandwidth are usually refused a license. For example, the FCC requires at least 4.5 b/Hz of bit packing. Nortel, among other vendors, have fielded radio systems with 9 b/Hz of bit packing in a quadrature amplitude modulated (QAM) configuration (i.e., 512-QAM).

Symbol rate is often confused with the baud. One might say that the symbol rate to the line of the V.29 modem is 2400 symbols/s, where each symbol carries 4 bits of information. However, "symbol" is more used in the radio world. In any case, I think its use as such brings about an ambiguity in terminology. Let's leave the symbol to the forward error correction (FEC) people where the output of a coder is measured in (coded) symbols per second (Fig. 4). Often the output of such a coder is fed to an *M*-ary modulator, and its transmission rate is also measured in symbols per second. This is confusing and ambiguous. Let's measure the output modulation rate of the modulator in bauds, and allow the output of the coder to be measured in symbols per second. The way I would recommend that these terms be defined is expressed in Fig. 4.



■ **Figure 4.** *A digital transmission system illustrating the use of the terms bits per second (b/s), coded symbols per second, and bauds.*

THE CONVERSE OF BIT PACKING

Spread spectrum systems do just the opposite of bit packing; they bit spread. Instead of bits per hertz in bit packing, spread spectrum could be typified in hertz per bit. The performance of these systems is given in processing gain: the more "hertz per bit," the greater the processing gain. For example, if we were to transmit 1 kb/s of data in the binary domain, it would require between 1 and 2 kHz of bandwidth, depending on filter rolloff, using conventional modulation such as PSK. If the data bitstream is spread over 2 MHz, the processing gain, which is the spread bandwidth (in hertz) divided by the data rate in bits per second, would be $2 \times 10^6/1000$, or 2000. In decibels, this is 33 dB of processing gain.

There are many advantages to using spread spectrum modulation in the world of radio, where there is a lot of interference and crowding, which is typical of the cellular and PCS environment. The greater the spread, the more immune the system is to interference. This is just one of the several advantages of using spread spectrum modulation. Another advantage is the number of active users supported per unit of bandwidth. Engelson and Hebart [12] state that spread spectrum modulation can accommodate better than 15 times the users per unit of bandwidth when compared to the North American Advanced Mobile Phone System (AMPS), which uses frequency modulation and 30 kHz channel spacing.

SUMMARY

In this brief review of some telecommunications terminology, we have aptly shown that it is dangerous to equate bit rate and bandwidth. If one carefully defines parameters, assuming as a first estimation one bit per hertz of bandwidth, then, if we wish to transmit 10 Mb/s, a bandwidth of 10 MHz would be required. The statement is clear and the writer is covered in this case.

However, one should remember that there are other definitions of bandwidth. Five of these have been listed. Those from the radio regulatory world may wish to use the 99 eprcent power containment definition. As clearly defined in [1], the baud is the measurement of signaling speed. In other words, it is a measure of transitions per second. The number of transitions per second determine the required bandwidth. Only in the binary domain is bits per second synonymous with modulation rate measured in bauds. Bits enter an FEC coder, and its output is coded symbols per second. The bandwidth of a transmission medium is determined by the medium itself. The data rate that bandwidth can support is determined by several factors such as the modulation waveform employed, the group delay (envelope delay) of the medium inside certain bandwidth constraints, signal level, noise, and type of modulation. The interchangeable use of bandwidth and data rate in bits per second should be deprecated.

REFERENCES

[1] *The New IEEE Standard Dictionary of Electrical and Electronics Terms*, 6th ed., IEEE Std 100-1996, IEEE, 1996.
[2] W. D. Reeve, *Subscriber Loop Signaling and Transmission Handbook — Analog*, IEEE Press, 1992.
[3] *Transmission Systems for Communications*, 5th ed., Bell Labs., Holmdel, NJ, 1982.
[4] R. L. Freeman, *Telecommunication System Engineering*, 3rd ed., New York: John Wiley & Sons, 1996.
[5] F. Amoroso, "The Bandwidth of Digital Data Signals," *IEEE Commun. Mag.*, Nov. 1980.
[6] B. Sklar, "A Structured Overview of Digital Communications — A Tutorial Review — Part I," *IEEE Commun. Mag.*, Aug. 1983.
[7] R. L. Freeman, *Practical Data Communications*, New York: John Wiley & Sons, 1995.
[8] H. Nyquist, "Certain Factors Affecting Telegraph Speed," *BSTJ*, vol. 3, Apr. 1924.
[9] H. Nyquist, "Certain Topics on Telegraph Transmission Theory," *Trans. AIEE*, vol. 47, Apr. 1928.
[10] C. E. Shannon, "Mathematical Theory of Communication," *BSTJ*, July–Oct. 1948.
[11] W. R. Bennett and J. R. Davey, *Data Transmission*, New York: McGraw-Hill, 1965.
[12] M. Engelson and J. Hebert, "Effective Characterization of CDMA Signals," *Wireless Rep.*, Jan. 1995.

BIOGRAPHY

ROGER FREEMAN (rogerf@futureone.com) is an independent consultant in telecommunications based in Scottsdale, Arizona. His consulting activities carry him worldwide, particularly to Spanish-speaking countries because he is bilingual. He has written five basic textbooks on telecommunications published by John Wiley & Sons, such as the 2350-page *Reference Manual for Telecommunication Engineering*, which is updated annually. He also gives seminars based on his textbooks at the University of Wisconsin–Madison. He was guest editor of the *IEEE Commmunications Magazine* issue on desert storm communications (Jan. 1992) and was secretary of the Spain section of the IEEE from 1972 to 1978. He holds baccalaureate and Master's degrees from New York University.