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Basic Review on wireless Communication for 5G

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Abstract- The number of telecommunications innovations grew rapidly during the last half of the 20th century. Currently there is widespread and growing use of cellular phones, cordless phones, digital satellite systems, and personal mobile radio networks. The electrical components of every communications system generate something called thermal noise. Noise typically appears like a random (unpredictable) signal added to our desired signal, and the noise increases with system temperature. This chapter began by examining signals and how to determine the bandwidth of signals, in order to ensure that the transmitted signals had a small enough bandwidth to fit within spectrum allocations. It was shown how analog signals can be sampled and quantized, and that there are size versus quality tradeoffs in this process. Next, channel effects are studied.

I. INTRODUCTION

The number of telecommunications innovations grew rapidly during the last half of the 20th century. Currently there is widespread and growing use of cellular phones, cordless phones, digital satellite systems, and personal mobile radio networks. Wireless communications occurs at many different frequencies, from underwater communication at extremely low frequencies on the order of tens or hundreds of Hertz, to infrared at 10^{14} Hertz. See Fig. 1 for a partial diagram of the radio frequency (RF) spectrum. In the United States the spectrum is allocated by the Federal Communications Commission (FCC).

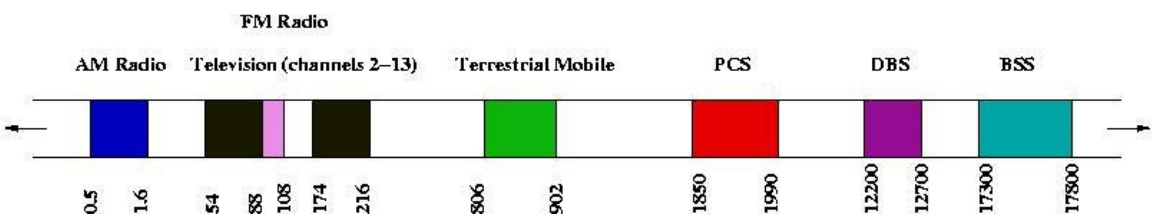


Fig. 1 A section of the RF spectrum showing some of the frequency assignments in MHz.

A significant development in telecommunications in the United States was the 1996 Telecommunications Act. This act was written in part to promote competition (telecommunications had hitherto been controlled mainly by a group of monopolies), promote integration of advanced services to all Americans and development of the underlying infrastructure. Furthermore, it created measures, such as a rating code, to deal with violence and obscenities, and laid out punishments for misuse, such as harassing phone calls, of the telecommunications systems[3].

The area of wireless communications will continue to grow for many reasons. People are becoming accustomed to immediate access to information wherever their locations, and technological improvements have made providing universal telecommunications access feasible. There currently is an expansion in the number of personal mobile radio networks that are the systems used by law enforcement groups, ambulance services, and on the floor of factories. The signals are meant to be relatively short-range and communication takes place on designated frequency ranges where they will not interfere with other applications such as wireless or mobile phones. In the near future there will be significant growth in wireless for the office, such as wireless local area networks and wireless private branch exchanges. New developments in personal communications systems (PCS) include integrated phone/paging/email/data transmission. Currently handheld units are offered by the major wireless industries with many of these features. These units range from cell phones with email capability, wireless pen tablets (low-end laptops without keyboards – interaction is via a pen), PDAs, and personal organizers, At the moment these have low-rate internet service on the order of 10 kbps, however speed and interconnectivity will be increased[6].

1.1. Communications Systems Overview

All of the systems mentioned previously, regardless of frequency or purpose, are communications systems. A communications system necessarily consists of three parts: a transmitter, a receiver, and a channel. The transmitter takes a signal, whether analog or digital, and formats it for transmission over the channel. A wireless channel can be water, air, or vacuum, and may contain obstructions such as buildings, terrestrial features, or planets, depending on the medium. The receiver captures the transmitted signal and performs signal processing, changing it from a form that can be transmitted over the channel into a form that can be viewed, heard, or stored. All of these system components introduce degradation to the transmitted signals; furthermore each system has a limit on the number of signals that can be transmitted. By carefully studying and compensating for the degradation caused by the system components, and by carefully designing the signal processing within a communications system, the number of signals that can be transmitted at one time can be maximized while the signals' degradation can be minimized. In the following sections the signals and system design tradeoffs are briefly considered [4].

II. INTRODUCTION TO SIGNALS

When we listen to radio or the telephone, or watch television, we are observing analog signals, that is, signals that are continuous in amplitude and time. Fig. 2 illustrates a segment of speech. From this figure we can see how the signal is able to take on any amplitude value within the range [-1,1].

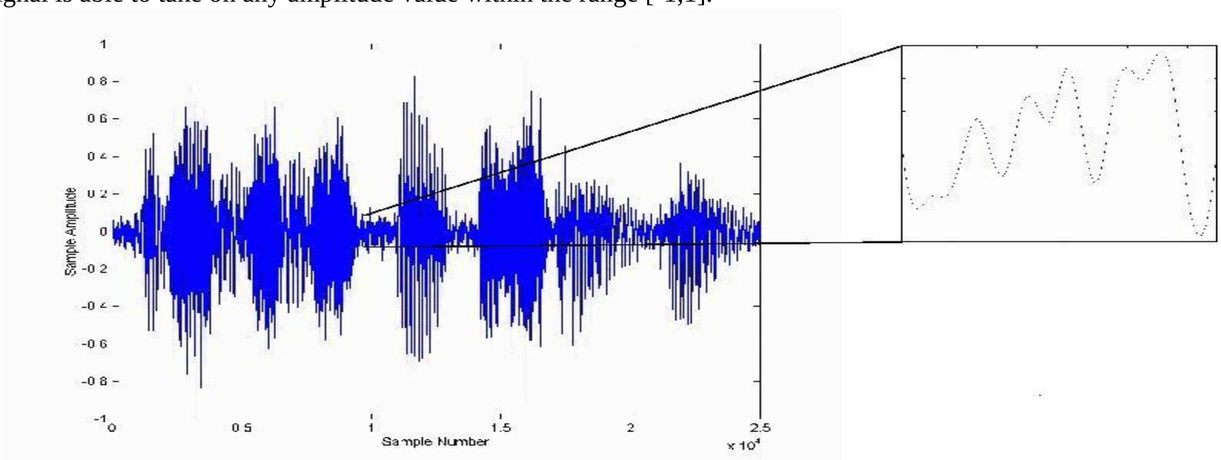


Fig.2 A sample of speech with a section extracted

As was mentioned in the previous section, one of the fundamental constraints to our transmission systems is the available bandwidth. The FCC only allocates a limited amount of bandwidth for each application, and no one is allowed to exceed his or her limitation. Therefore, we need a method of determining the bandwidth or frequency content of a signal; this bandwidth is measured in cycles per second, commonly called Hertz.

One of the greatest mathematical discoveries of the 19th century was made by Jean Baptiste Joseph Fourier, who determined that most aperiodic signals could be represented by summing their frequency components. That is, for most signals we are interested in the equation holds.

$$V(f) = \int_{-\infty}^{\infty} v(t) e^{-j2\pi ft} dt$$

This means that a time-domain signal, $v(t)$, such as our speech of Fig 2., can be represented by the different frequencies in it. In the frequency domain our signal is represented by $V(f)$. This is perhaps best illustrated by an example. Fig. 3 shows the Fourier transform of extracted segment of speech from Fig. 2. This figure demonstrates that the speech sample is composed of frequencies between 0 Hz and 16 kHz. This implies that if the channel or, correspondingly, the bandwidth allowance is greater than 16 kHz, then the signal can be transmitted over the channel and not interfere with any other transmitters. Thus, the Fourier transform is a very powerful tool in communications system design. Note that this is not a typical speech sample, since it has high-frequency noise; speech is typically bandlimited to 300-4000 Hz.

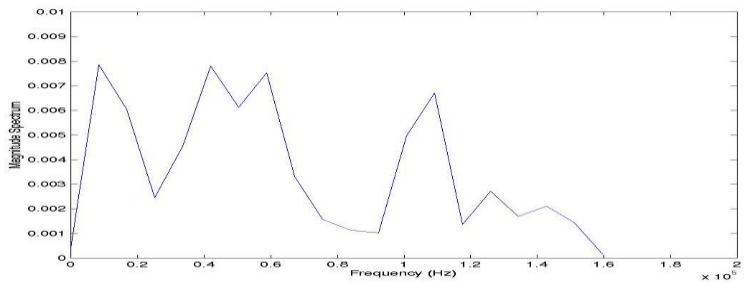
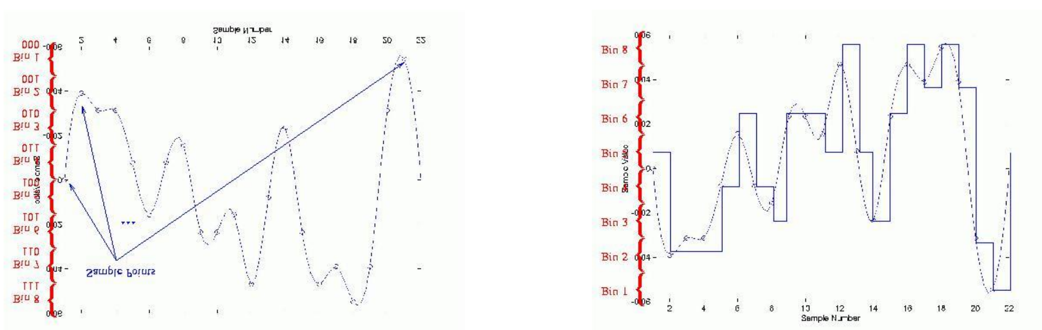


Fig 3. The discrete Fourier transform of the speech sample.

Rather than transmitting an analog signal, we may instead wish to transmit a digital signal. Digital signals are signals that are discrete in both time and frequency and may arise in many ways. For instance, to transmit information stored in the memory of computer such as an email, a stream of bits (1's and 0's), called a *bitstream*, is formed. We can also change our analog signals into digital signals through sampling and quantization. Fig. 4 illustrates the process of converting the voice signal of Fig. 2 to a digital signal. First, the signal is sampled periodically, that is every T_S seconds we record the amplitude of the signal. Nyquist proved that the sampling frequency, $f_S = \frac{1}{T_S}$, must be at least twice the maximum frequency in the signal. Typically, voice signals are sampled at a rate of 8400 samples/second. For compact-disk quality music, which is typically limited to the range 0 to 15 kHz, the sampling rate is 44,100 samples/second.



(a) Sampling and quantization of an analog signal.

(b) Reconstruction from sampled values.

Fig. 4 Sampling, quantization and reconstruction of a signal.

Note that the resulting signal is discrete in the time domain, but each sample can take on a continuous value. For instance, if we look at the first sample taken at time $t = 1$ the sample value $v(1)$ could be a number such as 0.01045972… with an infinite number of digits after the decimal point. To represent even this one sample as a series of bits would obviously require an infinite-length bitstream, this would involve much more computer memory or transmission bandwidth than we would be prepared to spend. Therefore we need a way to decrease the size of our numbers and we turn to quantization to represent each sample by a fixed (and usually small) number of bits. In Fig. 4(a) the vertical axis is divided up into 8 "bins". Each quantized value is assigned the midpoint of the bin. For example, any sample value falling in the range [0.0 ,0.019) would be assigned the value 0.0095. Then to represent the sample efficiently, the bins are labeled with binary sequences, and the sample falling in each bin is given the appropriate binary sequence. In this case, 3-bit sequences are employed, since it takes 3 bits to represent 8 numbers. Thus we see that the first sample $v(1)$ is assigned the value 0.0095 corresponding to bits 100 while the second sample is assigned the value –0.0395 corresponding to bits 001. Therefore, quantization of the first two samples results in the bitstream 100001.

Since the samples are assigned to bins, obviously the bin size, or number of bins will affect the quality of the quantized signal. Fig. 4(b) shows how a signal is reconstructed from quantized values. When the bits 100 are

received after transmission, in order to be fair, since it is impossible to know where in the range [0.0,0.019) the original sample fell, the value 0.0095 is assigned to this sample.
Therefore, the reconstructed signal, which is drawn in blue, becomes 0.0095 for the length of the sample interval T_S. It can be seen that with this bin size the reconstructed signal is an approximation to the original signal. To improve the quality of the reconstructed signal we could increase the number of bins, and hence increase the number of bits required to represent a sample. However, this increase in the number of bits/sample, although it results in a better reconstructed signal, requires more storage or more transmission bandwidth. A typical number of bits per sample for voice signals is 8, however, for compact-disk quality music 65,536 quantization levels are employed.
To further reduce the number of bits required to represent a signal, a compression scheme can be utilized. Common image and video compression standards are JPEG and MPEG; these are based on further quantization of signal components and are used in digital television. Speech and other signals rely on schemes such as linear prediction, which relies on estimation of signals and transmission of the difference between the true signals and their estimates. These types of compression are lossy; they discard information, and thus reduce the quality of the signal. However, when carefully employed, the degradation may be kept to a minimum, or even be made not perceptually apparent, and the bitstream size significantly reduced.

III. SIGNAL PROPAGATION AND CHANNEL EFFECTS

Besides the lack of readily available bandwidth and the number of users who desire access to wireless systems, the largest obstacle to building systems is that of noise and fading. This was seen in the early transmission experiments, the Morse code dots and dashes were hidden in noise, making long-distance transmission a challenge. Noise in car radios is a familiar phenomenon, as you drive away from a transmitter, the station becomes noisier until it finally drops out and all you can hear is static. This effect derives from the decreasing power in a received signal as the transmitter-receiver separation increases. The power at the receiver is governed by the following equation

$$P_R = P_T \frac{A_T A_R c^2}{\left(4\pi fd\right)^2} \quad , \tag{1}$$

where P_R and P_T are the received and transmitter power, respectively, A_R and A_T represent the amplification of the receiver and transmitter antennas, c is the speed of light $\left(3\times10^8\ m/s\right)$, f is the signal frequency, and d is the distance between the transmitter and receiver. We can see from this equation that the received power decreases with the square of the distance from the radio to the transmitting antenna.
The electrical components of every communications system generate something called thermal noise. Noise typically appears like a random (unpredictable) signal added to our desired signal, and the noise increases with system temperature. A picture of a clean signal, that is, our transmitted signal, and the signal with noise, $n(t)$, added to it is shown in Fig. 5. According to equation (1) as the receiver and transmitter separation grows the received signal power decreases. After a time the received power is so low that the noise becomes audible, and, if the separation is wide enough, the noise dominates, so that all you can hear is static. This static is an example of the noise in the system, it is entirely random so you cannot hear anything recognizable in it. Because it is unpredictable, it cannot easily be removed from the desired signal without degrading the desired signal.

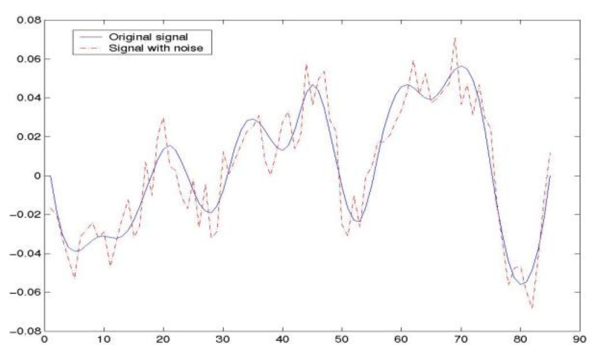


Fig. 5 A signal with noise.

In addition to noise, signals are often subject to fading. Signals can be reflected off of the ground, buildings, walls, trees or almost any object in their paths. One result of this is that, on average, the signal strength may decrease by a factor greater than the square of the distance. Furthermore, receivers at the same distance from the transmitter, but in different directions, may have greatly differing signal strengths. This phenomenon is accounted for by a path loss exponent, *n*, which is a number computed by many measurements in many areas and is the power to which the distance, *d*, in (1) is raised. It is further accounted for by a random number for each location at radius *d* from the transmitter. Thus, the power loss equation of (1) can be rewritten as

$$P_R = P_T \frac{A_T A_R c^2}{\left(4\pi f\right)^2 d^n} N \,. \tag{2}$$

In this equation, *N* is a noise factor, which is determined for different terrains and can capture some of the differences in signal strengths. The parameter n can take on values of 2 for free space loss (as in equation (1)), 4 for some urban cellular systems, and can range as high as 6 for intrabuilding communication. Naturally, this parameter will vary depending on whether the signals can penetrate walls and how many buildings or other obstacles there are in the neighborhood.

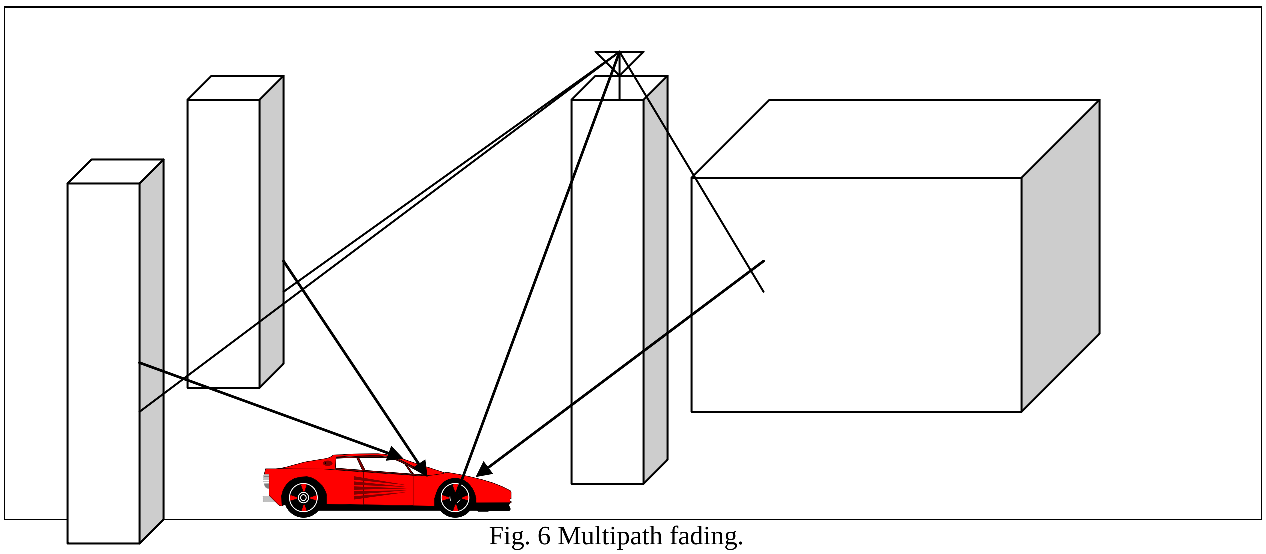


Fig. 6 Multipath fading.

Another problem called multipath adds challenge to the signal transmission. Multipath is illustrated in Fig. 6 where four signals are received at the transmitter. Each of the four has traveled a different path and is received at a different strength. Thus, the total received signal is the sum of these four signals

$$r(t) = \sum_{i=1}^{4} a_i v(t-\tau_i) e^{j\varphi_i} \tag{3}$$

Each signal on each path is delayed by time τ_i, and has amplitude and phase shifts a_i and φ_i, respectively. One can imagine the difficulty in extracting an unknown signal embedded in many others. Finally, movement of the mobile will affect the received signal by producing a change in frequency. This, as with noise is a familiar phenomenon; think of the wail of an ambulance approaching and then leaving, the frequency increases and then decreases. If a sinusoid is transmitted, the received frequency is the sum of the transmitted frequency and the Doppler shift

$$f_R = f_C + f_D \,, \tag{4}$$

where f_C is the transmitted sinusoid. Assuming the receiver is co-linear with the mobile the Doppler shift is given by $f_D = \dfrac{v f_C}{c}$ where *v* is the velocity of the mobile. As the velocity increases (the mobile moves toward the transmitter) the apparent frequency increases, as it moves away the frequency decreases. Since mobile velocities are rarely constant, this frequency can change quite a bit, making reception a difficult prospect.

Therefore, it is obvious that there are many challenges in system design. Transmitters must be closely spaced closely or use large enough powers so that receivers can overcome the inherent system noise. If there is multipath, fading and Doppler shift, these must be compensated for either with careful signal design or intelligent receivers. One method of protecting against poor channels is channel coding. Bits are added to the bitstream in a controlled manner, so that if noise or fading degrades some bits, these lost bits can be recovered from others. Because the size of the bitstream is increased, the bandwidth must be increased proportionally in order to maintain the transmission rate. Thus, there is another tradeoff between protection against channel errors and bandwidth required.

IV. MODULATION FOR ANALOG AND DIGITAL TRANSMISSION

In order to transmit a baseband signal, which is an analog signal composed of frequencies near 0 Hz, at radio frequencies we need to change or *modulate* a high-frequency carrier with our signal. There are two primary methods of analog modulation, amplitude modulation (AM) and frequency modulation (FM); this is where our radio transmission schemes take their names.

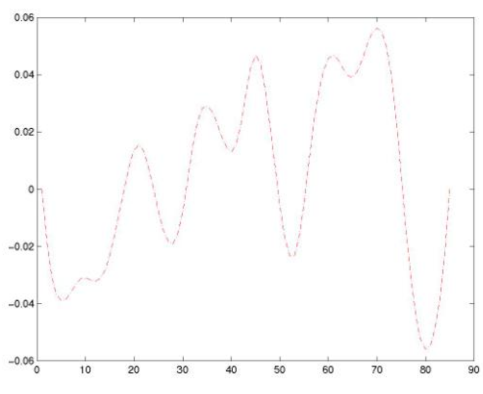
In AM we change the amplitude of a carrier by our message. For AM radio the carrier or sinusoidal wave has frequencies in the range of 540-1700 kHz. The equation for AM is given by

$$v_c(t) = A_C[1+\mu v(t)]\cos(2\pi f_c t) \tag{5}$$

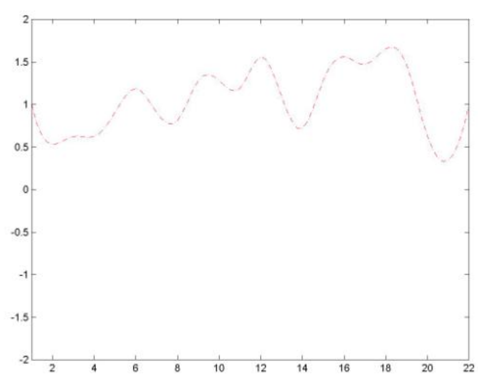
where $v_c(t)$ is the modulated signal to be transmitted, $A_c\cos(2\pi f_c t)$ is a carrier of amplitude A_c and frequency f_c, and the baseband analog signal, $v(t)$, is scaled by a modulation index μ. The construction of the AM signal is shown in Fig. 7. Fig. 7(a) shows the original signal, while Fig. 7(b) shows the signal scaled by the modulation index and shifted by 1. Fig. 7(c) shows the original carrier, which has a frequency much greater, typically by several orders of magnitude, than that of the baseband signal. Finally, Fig. 7 (d) shows the modulated signal. Observe how the message, the original baseband signal $v(t)$, is contained in the amplitude or envelope of the modulated signal $v_c(t)$. Frequency modulation is similar to amplitude modulation, except with FM the amplitude of the carrier remains constant, but the frequency changes with the message. The FM equation is

$$v_c(t) = A_C\cos(2\pi f_c t + \eta\int_{-\infty}^{t} v(\lambda)d\lambda) \tag{6}$$

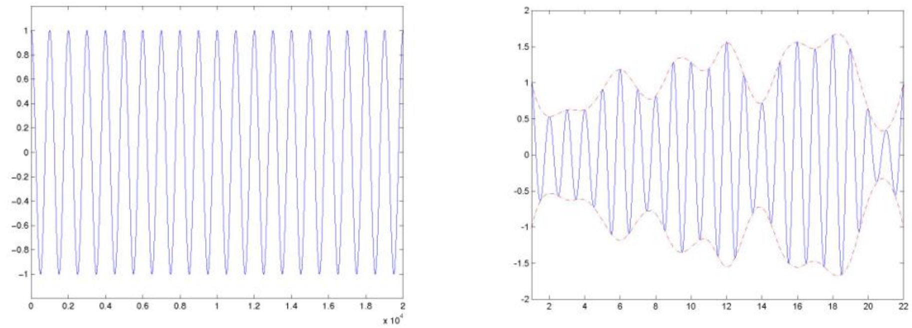
Now the integral of the message, scaled by a modulation index η, changes the phase or, correspondingly, the frequency of the carrier $A_c\cos(2\pi f_c t)$. Fig. 8 shows the message and the results of modulating the carrier. Observe how as the amplitude of the message increases the frequency of the carrier increases.



(a) Original signal.



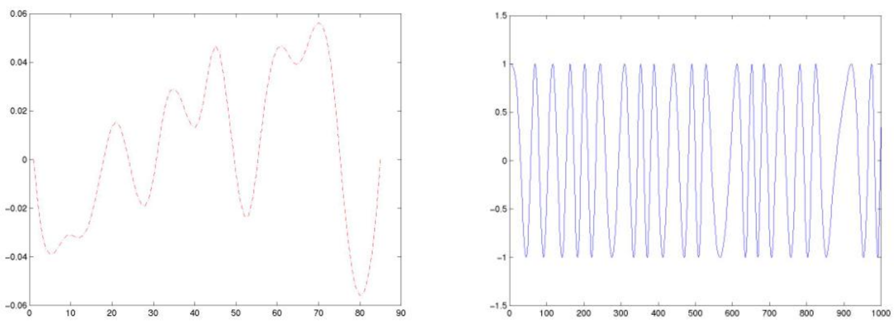
(b) Scaled and raised signal.



(c) Carrier

(d) AM signal with information embedded in the shape of the high-frequency carrier.

Fig. 7 Generation of an AM signal.



(a) Original Signal.

(b) FM signal.

Fig. 8 Generation of an FM signal.

4.1. Digital Modulation

Since bitstreams are not continuous signals, the AM or FM modulation schemes described above cannot be directly employed, and different ways to modulate a high-frequency carrier must be found. One method is binary phase shift keying (BPSK). While the name is complicated, building the modulated waveform is actually a very simple process. To transmit a 1 the signal $v_1(t) = \cos\left(2\pi f_C t\right)$ is transmitted for time T_b, while to send a 0 the signal $v_2(t) = \cos\left(2\pi f_C t + \pi\right)$ would be transmitted. Note that these two signals are $180°$ out of phase with each other. In Fig. 9 the BPSK signal corresponding to the bitstream 10110 is shown. Observe that there is a phase change at every bit change, these changes occur at times $t = 1000, 2000, \text{ and } 4000$ seconds.

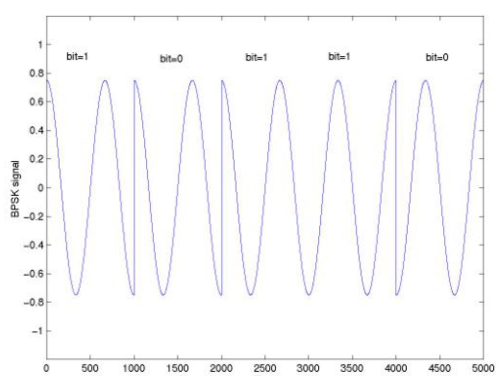


Fig. 9 A BPSK waveform.

In order to determine which bit was transmitted in time interval $nT_b \le t < (n+1)T_b$ all we have to do is to detect the phase of the received signal in that time interval. A more common method of transmitting bits is to take two bits at a time from the bitstream and use four-level phase shift keying.

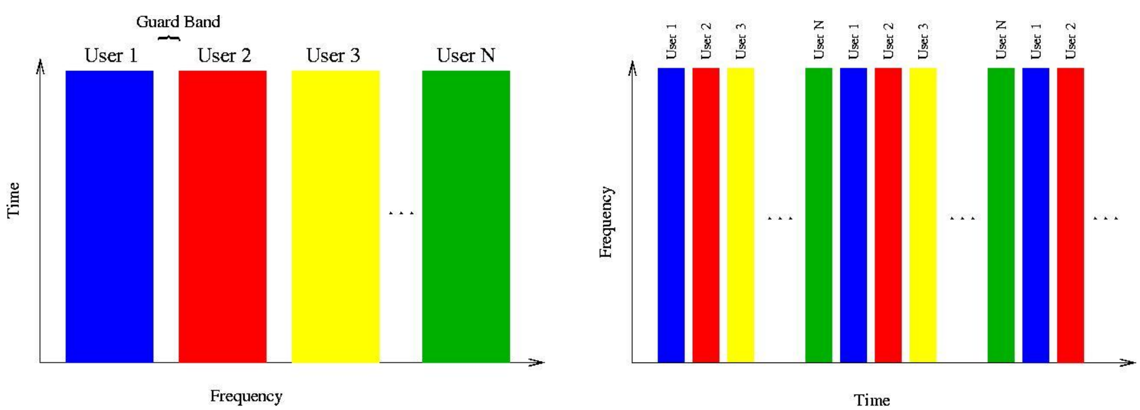
Variations on this method are used in many of today's digital systems today. Since four signals are required to represent all of the possible combinations of two bits, we then assign each combination the associated signal, Again, we detect which of the four signals was sent by determining the phase of the received signal.

$$v_i(t) = \cos\left(2\pi f_C t + \frac{\pi i}{4}\right); \quad i = 1 \ldots 4 \tag{7}$$

Naturally, there are other methods of transmitting bits, we could instead choose between two signals of different frequencies or amplitudes to represent our two bits. Another method is to use a combination of phase-shift keying and amplitude modulation; the information is then contained in both the amplitude and phase of the signal. The different modulation schemes are selected on the basis of their robustness in the face of noise and fading, the power required to transmit each and the complexity of the hardware required for transmission.

V. MULTIPLE ACCESS

Until this point we have discussed how to create a signal for one user, but the available bandwidth must be shared between many users. There are many methods of doing so; two of these methods, frequency division multiple access (FDMA) and time division multiple access (TDMA), are what is termed controlled multiple access, other methods such as code division multiple access (CDMA) and carrier-sense multiple access (CSMA) effectively permit users to access the channel whenever desired, under certain constraints.



(a) FDMA frequency division. (b) TDMA time division.

Fig. 10 Division of frequency or time in two multiple-access schemes.

Frequency division multiple access implies splitting the available spectrum between the users. This the method broadcast radio and television stations employ, each station is assigned a band (a certain range of frequencies in which they can transmit) and there is a short band in between the limits of each station called a *guard band*. The guard band is used to protect against flaws in the system such as carrier drift. A diagram of this is shown in Fig. 10(a). In the U.S. analog cellular phone standard (AMPS) the channels are 30 kHz wide and there are a total of 832 channels in the system, each having a forward and a reverse link.

For the TDMA system each user has control of the total channel bandwidth for a short amount of time, then the channel is handed off to the next user. Each waiting user has a turn and then control is returned back to the first user. A diagram of a TDMA system is shown in Fig. 10(b). TDMA is used in the European cellular phone standard Global System for Mobile (GSM). Each channel supports 8 users; each user is allowed to transmit for 577 microseconds before it is the next user's turn.

Code division multiple access is a hybrid system, which allows *all* users to occupy the same bandwidth and time simultaneously. Essentially, everyone transmits at the same time; signals are differentiated at the receiver because they are orthogonal to each other.

The orthogonality and knowledge of characteristics of the orthogonal signal make it possible to extract one user's signal from the entire transmission. The CDMA detection process can be envisioned as a noisy room. Without concentrating it seems as if what you hear is simply an unintelligible combination of sounds. However, if you can

focus on a familiar voice, the words from this voice start to become distinguishable, and you can block out the background noise in the room.
CSMA is a method used in many wireline communications that can also be used for digital wireless communications. Each user's voice or data is broken into packets. The user then listens in on the channel in order to determine if anyone is transmitting. If the channel is unoccupied, then the user transmits.
However, a collision occurs if two more users attempt to transmit at the same time. The receivers detect the collision and the users must retransmit their information at a later time. This system is only efficient if there is not a strict time constraint on the data, or if there are few users who wish to simultaneously transmit.

VI. CELLULAR SYSTEMS, FREQUENCY REUSE AND WIRELESS NETWORKS

Even with the multiple access schemes mentioned previously, only a small number of users could be handled at the same time. Certainly, one system would not be able to handle an entire city, perhaps not even an entire building. Therefore, in order to accommodate more than a small number of users, space is divided into cells as illustrated in Fig. 11. Each cell contains a basestation that handles all mobiles within the boundaries of the cell. Cells do not necessarily have to be of the same size. In rural areas cells are large, with radii of kilometers. On the other hand, consider a building such as the Philadelphia Convention Center. During conventions there are expected to be thousands of people in the building; they may all access their mobile phones at the same time, for instance, after the conclusion of a seminar. Thus, there may be tens or hundreds of microcells, on the order of a few square meters within a building such as this. Each cell is assigned its own frequency range, and neighboring cells will not be assigned the same range. Thus, the spectrum is shared in a cellular system much as it is in the radio and television system today. Cities which are widely enough separated, for instance, Philadelphia and New York, have radio and television stations at the same frequencies, the frequency spectrum is re-used in space. In a cellular system if cells which are assigned the same frequency range are widely separated, then there will be little inter-cell interference. However, if there is to be wide cell separation, there must be a large number of frequency ranges to assign, and thus each range is small. This limits the number of calls that can be handled in a cell, consider FDMA as an example.
Fig. 11 illustrates the interconnections of basestations, the transmitters/receivers of the cells. Three cells of an *n*-cell system are shown. The basestations are connected via a landline or microwave link to the mobile telephone switching office (MTSO), which controls all of the calls in this *n*-cell region. The MTSO also routes calls to the public telephone system, the traditional wireline system.

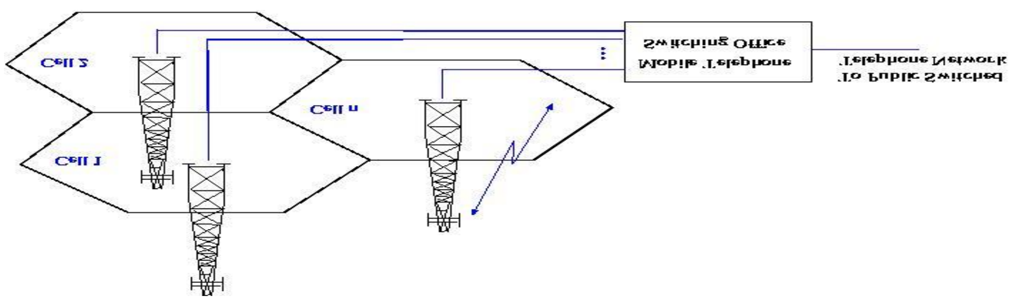


Fig. 11 An example of cell and basestation layout.

There are many important considerations in the design of cellular systems, ranging from cell placement, to switching users between basestations as they move from cell to cell, to providing service to users when they are roaming outside their provider's network. Each of these is discussed briefly below. Cell layout and system development is a difficult proposition. To determine the size of cells required in a particular area, precise traffic models, models of the number of users at a given time, must be developed. A provider would like to be able to provide service to all users within an area, but the cost of erecting basestations and expanding an existing network is large. The space for a basestation must be bought or leased and it must be connected, either through a wireline connection or a high-speed microwave or optical link, to the network. Furthermore, space is not always available fo basestations. In the U.S. there are many service providers, in each area each provider establishes its own networks. There is continual competition for the best basestation locations, typically on high buildings or hills. Often there is resistance from communities to new basestations in their neighborhoods, either from dislike of the aesthetics of the towers or from concerns about the effect of radio waves on human health. In a mobile system, since the users can travel between cells, the number of users within a cell at a given time can never be known exactly. However, given the number of slots available in a cell, C, and the average usage within a cell of a selected size, T, the grade of service providable can be calculated from

$$\Pr\{blocking\} = \frac{T^C / C!}{\sum_{k=1}^{C} T^k / k!} \tag{8}$$

This is the probability that a call will be blocked, that the entire channel will be entirely occupied by other users when the call is placed. If this number is greater than users will tolerate, the parameters such as the cell size must be adjusted to compensate, balancing the cost of cell placement with the desired grade of service. The switching of a call from one basestation to another as a user travels through cells is termed a *handoff*. Often, just turning a corner and moving out of sight of the basestation can decrease the mobile received power enough to require a handoff to a stronger, yet more distant basestation. There are four types of handoffs varying on whether the handoff is controlled by the mobile or the basestation or network. There are advantages to these, first, if the mobile makes the decisions, it can react quickly. However, if the network or basestation makes the decisions the delay can be long, since the basestation must judge the received signal, therefore, there are advantages and disadvantages to all types of handoffs. The main requirement of handoffs is that they should be seamless, that is, the user should have no awareness of the switch between basestations.

Roaming is placing a call from outside a home area, typically the roaming should be seamless, and invisible to the user, that is, the user should have no idea that the call is being placed through a system other than the home system, with the possible exception of a roaming light or indicator. Therefore, systems must have roaming agreements in place and be compatible; it must be possible to identify a user's home system as the call is being placed and the two systems must be able to exchange information for billing purposes.

There are many competing wireless systems and services in the U.S., some are all analog, some are all digital, while others are hybrid systems which operate at times on either digital or analog. There are also competing standards, which use different multiple-access and modulation schemes. Therefore, if a user would like to change service providers he or she must purchase a new phone along with the service. Furthermore, many of the European and Asian standards are different from the U.S. standards, thus phones for U.S. systems may not work abroad. There is some movement in the U.S. to adopt the European mobile phone standard; this will lay the basis for extension of global service.Another increasingly employed solution to worldwide coverage is to manufacture a multi-mode phone, which is able to detect the type of system of the local provider and adjust transmission accordingly.

Up until this point we have considered mainly the mobile telephone networks; however, paging, PCS and other mobile networks operate on the same principles. Furthermore, although mobile wireless networks dominate the industry, there is a growing market for fixed wireless networks. This class of networks may have the same cellular layout as a network for mobile communications, but has both fixed basestations and users. A simple example is an infrared wireless link from computers to printers in an office, while a much more complex system could involve providing wireless internet access to a community. The reason for the growth of interest in fixed wireless systems is the cost of installing cable or fiber in established areas, and the ease of relocation possible with wireless. Additionally, a fixed wireless network does not have the problems of time-varying fading or Doppler shift and does not require the extensive processing for handoffs and roaming as would a mobile network, and thus is easier to implement. In the future there will most likely be an increased number of wireless systems, but also a move to standardize more so that these systems can intercommunicate. With an ever-increasing number of services and a desire for seamless handoffs between providers, developing the next generation of wireless networks will present significant challenges.

VII. CONCLUSIONS

It is apparent that the design of communications systems is a complex process with a large number of tradeoffs to deal with the limitations of channels and equipment. This chapter began by examining signals and how to determine the bandwidth of signals, in order to ensure that the transmitted signals had a small enough bandwidth to fit within spectrum allocations. It was shown how analog signals can be sampled and quantized, and that there are size versus quality tradeoffs in this process. Next, channel effects are studied. It was seen that there is inherent noise in systems, but that this noise can be overcome by increasing the power at the transmitter. On the other hand, large transmitter power requires larger batteries, a stringent constraint in applications such as mobile telephony. In wireless channels there can be multipath or time-varying fading and Doppler shift to complicate reception.

Having discussed single signals, we then considered how to share channels and examined four primary methods. Next, the further sharing of the spectrum was considered with an examination of cellular systems and frequency re-use. Several of the inherent challenges were discussed, it was seen that systems must be carefully developed and

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synchronized in order to provide for all users and to provide the services desired. Finally, as a case study, the Iridium system was discussed.

Even with the enormous growth in wireless communications systems within the past few decades, there are constantly new advancements in research and development. Currently, there are many concerns about the security and reliability of wireless systems, challenges which are important but which are not addressed here.

There is certainly much more to be done, frequencies to be explored and systems to be developed, that a quotation by Marconi appropriately summarizes the current state of wireless communications

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