Chapter 14: Telephony and Cellular Networks

Chapter 14 Objectives

At the conclusion of this chapter, the reader will be able to:

- Describe the overall topology of the U.S. telephone network.
- List the electrical characteristics of an analog POTS subscriber local loop.
- Describe the operation and characteristics of a typical analog CPE
- Describe the DTMF signaling standard.
- Describe the digital signaling standards used in the telephone system.
- Describe the operation of digital traffic switches.
- Compare and contrast AMPS and PCS service characteristics.
- Describe how subscriber signals are carried through a PCS network.
- Given a frequency reuse map for a PCS market, identify the groups of frequencies utilized by each cell in the market.
- Describe the operation of DSL/ADSL subscriber services.
- Describe a typical installation for VoIP equipment.

"Mr. Watson, come here, I want you." These historic words were the first speech to be successfully transmitted through wires, uttered by Alexander Graham Bell to his assistant in March of 1876. It was the beginning of the second wave of the telecommunications revolution, started by the invention of the telegraph in 1833. Now a basic part of modern life, the telephone system is one of the most creative and complex achievements of humankind. Over its lifetime, it has directly influenced and benefited from many advancements in electronic technology. The modern telephone system uses analog and digital circuitry, computers, advanced signal encoding, and of course a healthy dose of wireless technology. Let's see how it all works together.

14-1 The System View

In the earliest days of the telephone, no more than a few hundred telephone sets existed. Customers of Bell's new telephone company usually purchased telephones in pairs, and a single dedicated wire was strung between the phones (the earth provided the ground return path). Such an arrangement was suitable for local point-to-point communications, but hardly practical for anything else. Suppose that a community has 1000 phones, each of which must be capable of talking to any of the other phones. That might mean that each residence would require 999 lines for connecting to each of the other phones in the network. This would not do at all! Furthermore, what if someone wanted to call a relative in another city? The problem just became a lot worse. At the turn of the twentieth century, cities were becoming "blackened with wires" because phone lines were being strung from every available rooftop, pole, and building. Many inventors and engineers would work over a period of decades to provide working solutions to this fundamental problem.

The modern telephone system has evolved over a century from this crude beginning, and it continues to change. It has several important features. First, the national telephone system is a hierarchical arrangement of calling centers, as shown in Figure 14-1. The word *hierarchy* means a system ranking specific kinds of things, with the most important (and usually least numerous) things at the top, and others below. Most hierarchies look like upside-down trees. The higher levels of this system serve large geographic areas, such as the class 1 regional centers in the figure. The lower levels serve a small area, such as a section of a city.

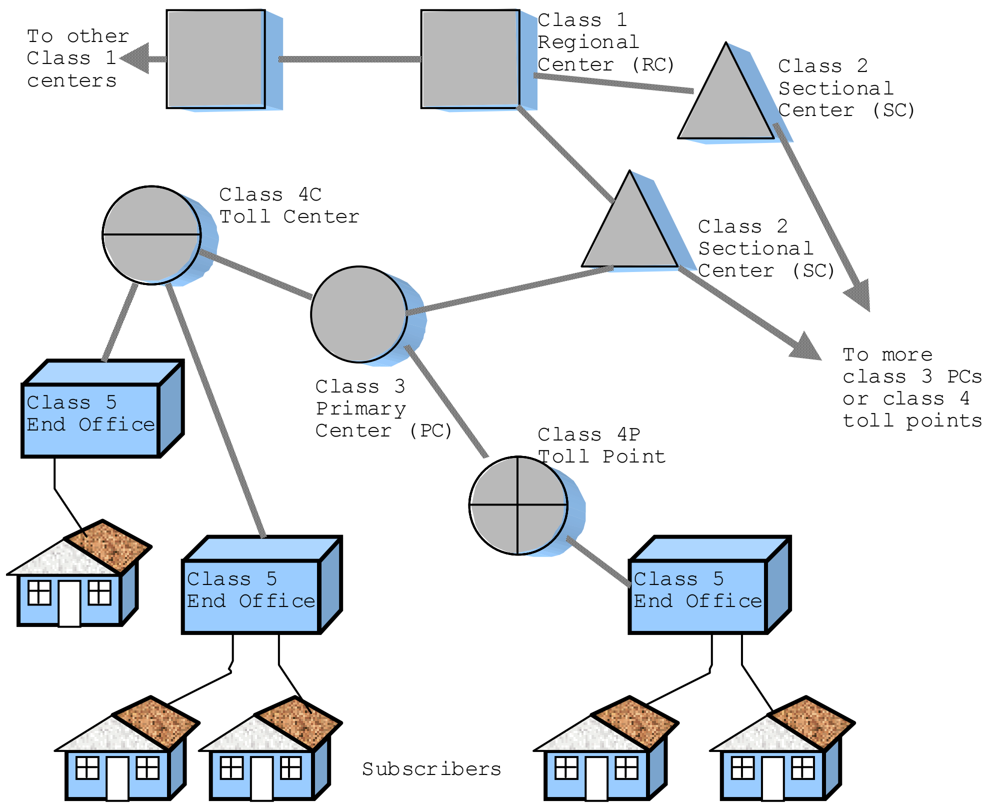

Figure 14-1: The National Telephone System Plan

The second important characteristic of the phone system is that it is divided into regions. This has become a little more difficult to understand since the government-mandated breakup of the Bell system in 1984. Each of the Bell operating companies (sometimes referred to as the "RBOCs," for regional Bell operating companies) has a specific geographic area that they are permitted to provide service in. This is called a local access and transport area (LATA). LATA boundaries are defined in the Modified Final Judgement (MFJ) handed down by the FCC in 1984. A region may cover a state or parts of several states.

Third, the telephone system is by and large completely automated. No human intervention is required for more than 99% of any local or long distance calls anywhere in the country. Finally, the telephone network uses digital signaling to carry almost all conversations. The only remaining analog part in the system is the customer loop (the copper connection between a residence and the central office) and CPE (customer premises equipment). Prior to the use of digital speech transmission, analog facilities were needed to transmit conversations from point to point on the network. Analog signals weaken and accumulate noise over distance, and must pass through repeater-amplifiers every 10 to 50 miles, or they'll become too small to be useable. Every stage of processing degrades an analog signal, so that after many miles and "repeater hops" it no longer sounds very good! Because of the use of digital coding, speech signals are no longer degraded as they are repeated through the telephone network. Also, replacement of copper by fiber optics has also increased the distance between repeaters. We will study how speech is represented by digital codes in this chapter.

Order of the Telephone System: How a Call is Routed

Figure 14-1 tells us a lot about how calls get from point to point in the system. Normally, calls originate from a subscriber connected to one of the central offices (sometimes called *end offices*). This office directly serves customers -- each residence has a copper twisted-pair line leading to the central office. A central office is designed to handle approximately 4,000 to 10,000 customer lines. Calls within the service area of the central office can be handled directly without any intervention from the levels above. Figure 14-2 is a typical suburban central office. They are deliberately designed to blend in with the surroundings (the phone companies would rather not have everyone be too aware of them) and typically are fortified against natural disaster and intrusion. Most have no windows but are otherwise unremarkable in appearance. There may be *no* staff at a central office unless it is a very large installation.

The switching equipment in the central office (CO) examines the number dialed by the customer. In particular, it examines the first four digits of the number closely. If the first digit is "0" or "1," then the call will likely have to be handed off to a *toll center* or *toll point*. Otherwise, the first three digits must be a *prefix* within the local calling area. Each central office serves a small number of prefixes. If the destination prefix is being served by the originating CO, then the call can be routed directly to the customer at the four-digit *station number* specified in the telephone number.

Suppose that the number is not long distance, but instead destined for a customer who lives across town. In this case, the call may be routed through a class 4 center, but more likely in a large metropolitan area, the call will be routed directly to the correct central office.

For out of area calls, the call must be routed up higher in the hierarchy. Between two areas that are geographically adjacent, the call may need to pass through a class 3 primary center. For larger distances, the call may need to "hop" sections, which may pass it through a class 2 sectional center, or finally, through a class 1 regional center. Notice that all traffic can't pass through the class 1 centers; doing so would create a bottleneck. Instead, the system is linked at several levels, so that a call bubbles only high enough through the system to make it to the destination.

Figure 14-2: A Central Office in South Kansas City

The Last Mile

The copper wire connecting a subscriber to an end office is sometimes called the *last mile* of the telephone system. It is highly valuable real estate, for it's usually the only way of connecting to a customer. Incumbent local exchange carriers (ILECs) usually own the last mile, and only recently have been forced by government regulations to share it with other carriers. An ILEC is the telephone company that was there to begin with and owns the copper wire leading to the customers in the region. This copper is sometimes called the *last mile* of the telephone system. Accessing the customer premises is often called the *last mile problem* in telecom circles. In some areas cable companies have begun offering telephone service, in many cases sharing the coaxial cable carrying TV signal with the analog telephone signal.

Section Checkpoint

14-1 What was the primary difficulty with Bell's original telephone system?
14- 2 List several characteristics of the modern telephone system, and explain each one.
14-3 Why is the hierarchical organization of Figure 14-1 necessary?
14-4 Explain how a call travels from house-to-house within a neighborhood, and contrast this with a call across the state.
14-5 What is the last mile problem?

14-2 The Local Loop: Operation, Signaling, and Telephone Circuitry

The local loop is the connection between a telephone subscriber and the central office. It includes the customer premises equipment (CPE), also known as the telephone set, and the equipment at the end office. With the exception of wireless calls (such as PCS), everything begins and ends here! The electrical characteristics of the local loop are standardized; many of these standard values date back to the beginning of the Bell system. Figure 14-3 is a simplified schematic of the local loop.

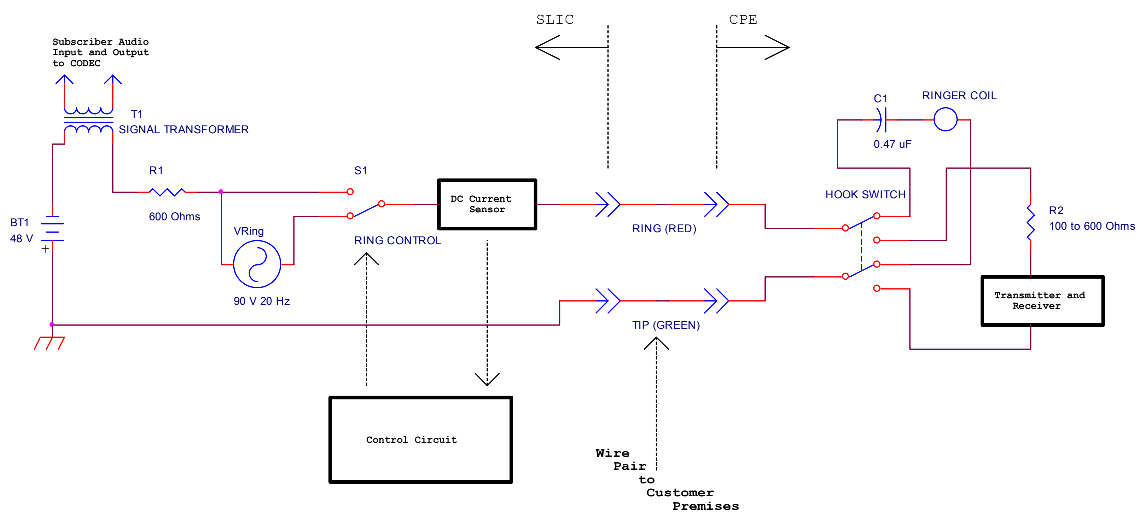

Figure 14-3: The Local Loop Connects the Customer Premises to the End Office

Local Loop Components

The components in the figure provide minimal analog telephone service, called *POTS* (plain old telephone service) in the industry. There are two primary parts in the local loop, the subscriber line interface card (SLIC) at the central office, and the CPE. These are connected together by the wire pair leading to the customer's residence. Although there are usually four wires at a telephone jack in a home, only two of the wires are usually active. The red wire, also called the *ring*, carries the voice and ringing signals, while the green wire, called the *tip*, provides the ground return. Occasionally the black and yellow wires are used for a second phone line connection.

The subscriber line interface card controls the electrical activity on the loop. It is responsible for receiving and sending the voice signals on the customer's line, ringing the customer telephone when a call is incoming, and detecting when the CPE is taken off-hook when the customer is beginning to place an outgoing call. The SLIC routes the status tones (dial tone, busy signal, and so on) to the customer and also accepts the dialing information from the CPE (passing it onto the telephone network). Finally, since most telephone signal switching is now digital, the SLIC is also responsible for converting the customer's outgoing voice signal into digital for transmission through the network, and converting the incoming digital voice signal from the telephone switching network back into analog so that it can be reproduced by the customer's telephone set. Most SLICs service eight customer loops simultaneously. Figure 14-4 is a peek inside an end office, showing the array of subscriber cards (plus other equipment) in the equipment rack. This end office is in a small town and probably wins the "messiest installation" award!

The telephone set or CPE is a fairly simple circuit (we'll look at it in more detail shortly). It consists of a hookswitch, ringer, transmitter, and receiver. The hookswitch disconnects the transmitter and receiver when the unit is on-hook, and connects the ringer. The ringer requires 90 volts AC, 10 mA at 20 Hz to operate, and is coupled to the line through capacitor C_1. This is important; the capacitor blocks DC current. *The end office detects an off-hook phone because it draws a DC current*. An on-hook phone can't draw a DC current because of C_1. When the handset is picked up the hookswitch is activated and connects the transmitter-receiver circuit onto the line. The transmitter and receiver draw a small DC current (usually 10 to 20 mA). The subscriber line interface card at the end office sees this current and determines that the unit is off-hook. Both incoming and outgoing voice signals share the same pair of wires; a circuit called a *line hybrid* in the telephone set helps to keep them separate.

Figure 14-4: Equipment Rack with SLICs Inside an End Office

Loop Operation

The circuit of Figure 14-3 operates on -48 V DC. This is a fairly standard value in telephony. This voltage is usually supplied by banks of batteries in a central office. These batteries are kept under continual charge, of course. Because batteries supply the operating voltages, momentary power interruptions don't disrupt the telephone service. (For longer disruptions, diesel generators are usually on site to keep the batteries charged.) The loop current is limited by resistor R_1, which is usually between 400 and 1200 ohms. The DC loop current is needed to operate the customer telephone circuits. An important part of the SLIC is the off-hook detection circuit, which is provided by the DC current sensor in the figure. When the customer's phone is on the hook it can draw no DC current from the line. The hookswitch in Figure 14-3 is shown in the on-hook condition, which connects the ringer and DC blocking capacitor into the circuit. Switch S_1 is in the "up" position in the figure so that no ringing voltage is present. Suppose that the customer decides to make an outgoing call. He or she picks up the handset, which moves the hookswitch to the opposite position. DC current now flows through the current sensor on the subscriber interface card. The control circuitry thus senses that a new call is starting, and a dial tone (a low-level AC signal) is sent onto the customer's line through signal transformer T_1. The customer then hears the dial tone and can begin the call.

A similar process occurs during an incoming call. The control circuitry detects the presence of the incoming call and moves switch S_1 into the position shown, which connects the 90 volt AC ringing signal onto the line. The DC level is almost *always* present on the line so that the end office equipment can continually detect the state of the customer's phone. Note that the 90 volt AC signal appears in *series* with the -48 volt DC bias so that the DC current sensor can detect when the customer's phone is answered, even while it is ringing. The ringing signal is turned off immediately when the off-hook condition is detected; then the circuit is ready for carrying the conversation. If at any time the customer hangs up the phone, the DC current will be interrupted, and the circuit will go back to the idle condition.

Example 14-1

Suppose that the customer short-circuits the *ring* and *tip* terminals at a residence. What is the maximum current that could flow through the telephone network, given the values in Figure 14-3?

Solution
With a short at the customer premises, R_1 will limit the total current. Ignoring the resistance of the copper wires (which may add 100 Ω or more to the total!), we get:

$$I = \frac{V}{R} = \frac{48V}{600\Omega} = \underline{\underline{80mA}}$$

Up to 80 mA could flow from the end office.

Example 14-2

What is the maximum DC power that a telephone set could get from the telephone network given the values in Figure 14-3?

Solution
To get draw maximum power from a source, connect a load resistance equal to the Thevenin resistance of the source (maximum power transfer theorem). Then use Ohm's law to calculate the power in the load. By inspection, $R_L = R_1 = 600\ \Omega$. The voltage that would appear across R_L would therefore be one-half of the 48 V DC provided at the end office, since R_1 and R_L form a voltage divider with equal devices. Therefore, the power delivered to the load would be:

$$P = \frac{V^2}{R_L} = \frac{(24\ V)^2}{600\Omega} = \underline{\underline{960\ mW}}$$

Less than a watt of DC power is available from the end office. This doesn't sound like much. However, consider that a central office serves thousands of users; the total power required is controlled by the number of off-hook telephones. In general, telephone facilities are not designed for 100% utilization, so if everyone takes their phone off-hook at the same time, only about 10 - 20% of the users will get a dial tone, and great stress will be placed on the end office power supplies!

Signaling on the Loop: Dialing a Number

Two primary methods are supported for customer dialing, *pulse* and *dual tone multifrequency* (DTMF). Pulse dialing dates back to the first automatic switching system, the Strowger switch, patented in 1891. DTMF, also known as *touch tone*, is supported by most subscriber line cards, and costs the telephone companies little (if anything) to support, but many phone companies still charge extra for DTMF service.

Pulse dialing relies on the operation of a mechanical telephone dial, as shown in Figure 14-5. The dial includes a return spring, a mechanical governor, and two sets of contacts. When the customer operates the dial, the *mute* contacts close to bypass the receiver so that he or she can't hear the dialing pulses. When the dial is released, it returns under power from the return spring, while the governor keeps the speed of the dial constant. The *dialing* contacts open and close the circuit according to the digit dialed; to dial "1," the circuit is opened and closed one time; to dial "2," the circuit is opened and closed twice; and so on. (Ten open-close cycles represent "0.") When the dial returns to the rest position, the mute contact opens to allow normal receiver operation again. In a modern telephone set the mechanical dial is replaced with a microcontroller and solid-state switch; the timing remains the same.

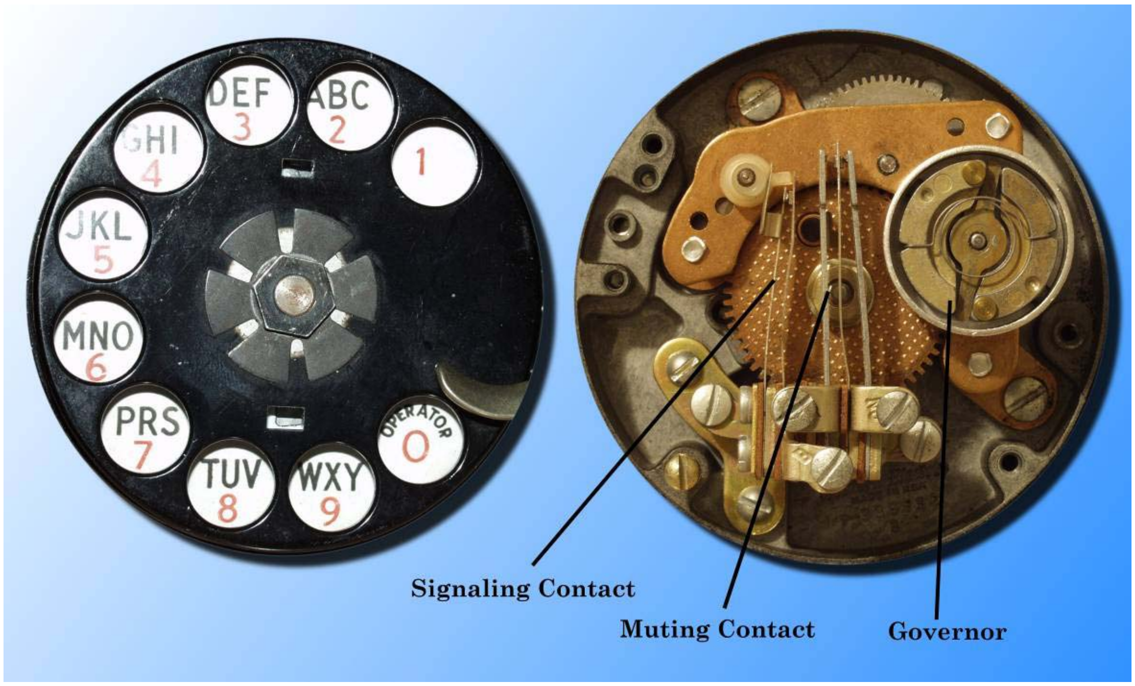

Figure 14-5: Mechanical Dialing Mechanism Details

Figure 14-6 shows the timing relationships for pulse dialing. There are a nominal 10 pulses per second (pps) in the United States, with a 40% duty cycle (the circuit is broken 60% of the time).

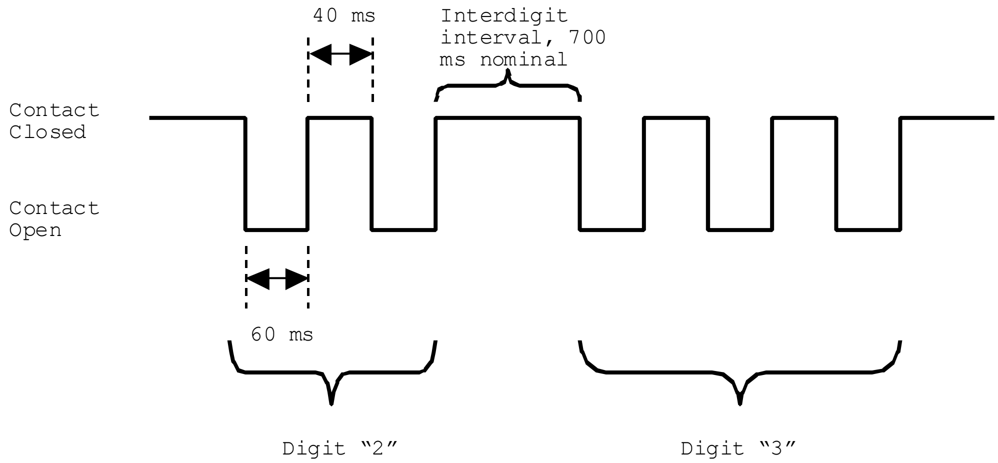

Figure 14-6: Pulse Dial Timing (United States)

DTMF or touch-tone signaling is much more popular since it allows quicker and easier dialing than pulse, and also allows access to more features (such as voice mail, banking by phone, and so on). In DTMF each digit is represented by two sine wave tones as shown in Figure 14-7. These frequencies were chosen carefully; none of them is a harmonic of the power line frequency (60 Hz) in the United States, and none of the tones is a harmonic of any of the other tones.

High-Group Frequencies (Hz)

Low-Group Frequencies	1209	1336	1477	1633
697	1	2	3	A
770	4	5	6	B
852	7	8	9	C
941	*	0	#	D

Figure 14-7: DTMF Tone System

To dial the digit "2" under DTMF, the frequencies 697 Hz and 1336 Hz are generated by the telephone set. The tolerance for error is ± 1.5% for acceptance of these tones at the end office; if either tone is out of tolerance, the keystroke is ignored by the telephone network. Modern telephone sets typically use a dedicated IC such as the National Semiconductor TP-5089 (usually under crystal control) to provide DTMF tones. Home telephones usually don't include the *extended column* of tones shown in Figure 14-7. These are used for special purposes within the telephone system, and occasionally for signaling for non-telephone applications (such as amateur radio).

DTMF tones must not only be accurate in frequency, but also in amplitude. Nominally, the transmitted level of the lower frequency group is -8 dBm into the 600 Ω impedance of the subscriber line, and the upper group's level is -6 dBm (boosted by 2 dB to since higher frequencies are attenuated more in the signal path to the end office). The difference in amplitude between the two DTMF tones transmitted by a CPE is referred to as *twist*. Excessive twist can cause one of the two DTMF tones to be rejected by the SLIC decoder, resulting in a dropped digit. Twist develops from uneven frequency response in the telephone system. Most DTMF decoders can tolerate up to about 6 dB of twist (2:1 voltage ratio of the two DTMF tones).

Example 14-3

What is the RMS and peak-to-peak voltage of a 0 dBm signal on a 600 Ω subscriber line?

Solution

Use Ohm's law to find the voltage. First, we must convert the dBm power level back into watts. By inspection, we know that 0 dBm is 1 mW of power, but we can also calculate the power by:

$$P = 1\ mW \times 10^{(dBm/10)} = 1\ mW \times 10^{(0/10)} = \underline{\underline{1\ mW}}$$

Since we know that $P = \dfrac{V^2}{R}$ by Ohm's law, we can solve for voltage:

$$V = \sqrt{PR} = \sqrt{(1mW)(600\Omega)} = \underline{\underline{775\ mV}}$$

In peak-to-peak units, this is:

$$V_{pp} = 2\sqrt{2} \times V_{RMS} = 2\sqrt{2} \times 775mV = \underline{\underline{2.19\ Vpp}}$$

The value of 0 dBm, or 2.19 Vpp, is also called "*0TLP*" in the telephone industry. It the maximum signal level that a customer device may place on the telephone line.

TIP: To accurately measure AC signal levels on a telephone line, you must either use an audio isolation transformer, or make sure that the test equipment is not connected to earth or power line ground (for example, use battery powered equipment). Otherwise, you'll be introducing a *ground loop* onto the line, with a great deal of power line noise! Commercial equipment is readily available to measure signal levels on POTS lines.

Telephone Line Frequency Response

During the development of frequency-division multiplexing techniques (which would allow a wire to carry more than one conversations), experiments by Bell Laboratories were performed to determine what frequencies were needed to carry a voice conversation. By restricting the bandwidth, more voice channels could be carried on a wire. The frequency response pattern of Figure 14-8 was adopted as a result. Dial-up telephone lines have this characteristic.

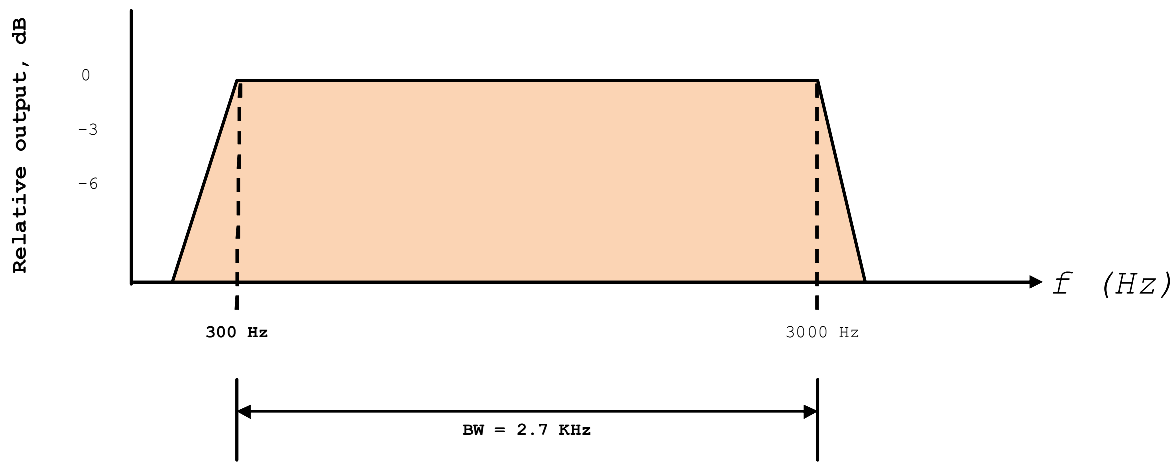

Figure 14-8: Ideal Frequency Response of a Standard Dial-Up Telephone Line

You'll notice that the frequency response of a telephone line is quite narrow compared to the full range of human hearing (20 Hz to 20 kHz). The rolloff at 300 Hz limits the bass presence of the voice, which adds very little to intelligibility. The high frequency limit is close to 3.3 kHz (with the upper rolloff starting at 3 kHz). The limited treble response reduces the crispness of the speech signal; most of the higher-frequency energy in speech comes from unvoiced consonants such as *f*, *s*, and so on. The frequency response of the telephone system is controlled by end-office equipment, primarily, though the transmission lines between a residence and end-office also have a high-frequency effect as well. In the modern digital phone network it is necessary to limit the high-frequency response prior to *sampling* the analog signal during analog-to-digital conversion, as we'll see in the next section.

In addition to the limits of frequency response, there are two types of delays a signal is subjected to as it passes through the telephone system. These are *propagation delay* and *group* delays. Propagation delay is caused by the distance a signal must travel to reach the destination. On a long distance connection this delay may amount to 10 ms or more. The telephone system employs echo cancellation equipment to prevent delayed long distance signals from echoing back and forth across a connection.

Group delays (also called envelope delays) are a *variation* of the delay with frequency. In other words, some passband frequencies may be passed faster or slower than others through the telephone system. This causes distortion of the received signal since the phase relationship of the various energies in the signal becomes incorrect, which means that the signal will have an incorrect shape. For voice signals, the distortion may hardly be noticeable, but for data communications, excessive group delays can corrupt transmitted data. Figure 14-9 shows the graph of group delay for a typical telephone connection. Note that the vertical axis shows the *differential* or *change* in delay with respect to the minimum delay, which is found at 1500 Hz in this example.

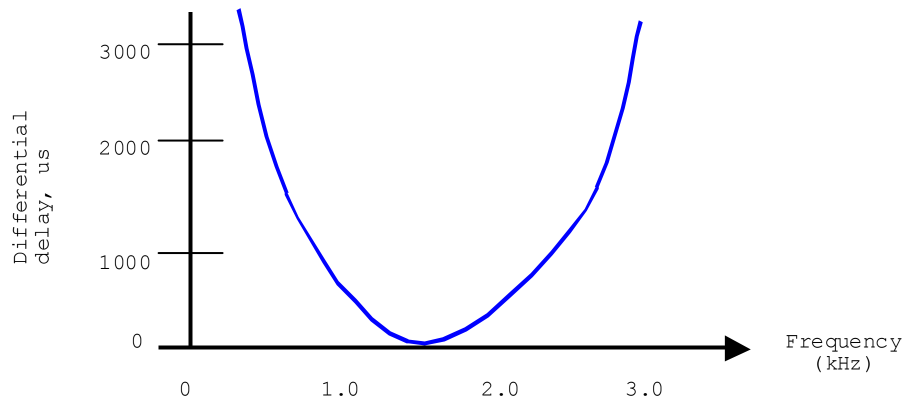

Figure 14-9: Group Delay Through a Telephone Connection

You can simulate the effect of group delay on a square wave by using the *Wavegen* program provided on the support website with this text. To do so, proceed as follows:

- Download and start the *Wavegen* program. If you have speakers on your PC, turn them on. You should hear a 1 kHz sine wave playing through them.
- Click the Show/Hide Scope Window button. A scope window will pop up to show you the 1 kHz sine wave being produced.
- Click the Square button. You'll see the scope window waveform change to an approximated square wave. (This could represent a digital data signal). The sliders representing the various harmonic energies (and their phase angles) will automatically move to show you the formula for the square wave.
- You can now adjust the phase angle of any frequency in the waveform (which for a square wave would be the first, third, or fifth harmonic *phase* sliders). This will introduce a phase shift into each signal, which is equivalent to the effect caused by envelope delay. You'll see the resulting waveform change as you adjust the sliders. See if you can hear the difference between the original and distorted waveforms!

Inside the Telephone Set

The electronics within a telephone set are relatively basic. A telephone set in its simplest form consists of a hookswitch, dialing mechanism, a receiver, and a transmitter. Figure 14-10 shows a simplified telephone set employing a mechanical pulse dialing mechanism.

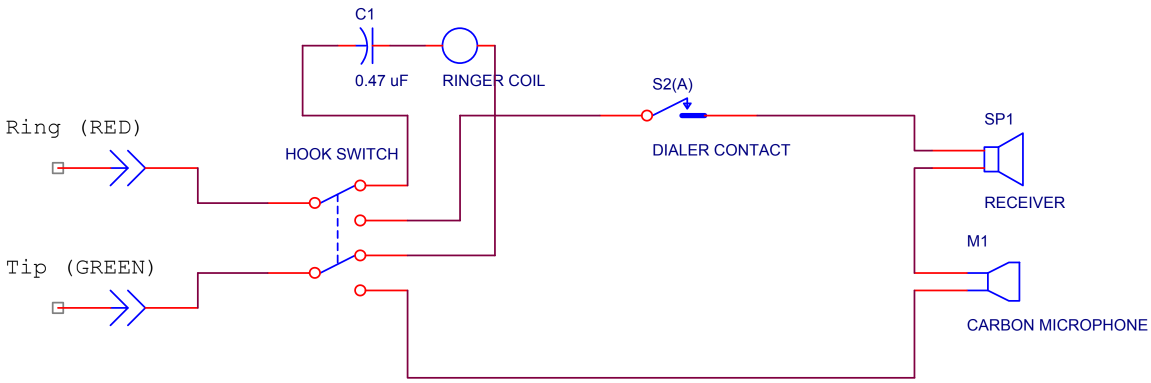

Figure 14-10: Simplified Telephone Set

The operation of the set in the figure is straightforward. When the phone is on-hook, the hookswitch connects the ringer coil and capacitor C_1 to the telephone line so that the incoming ring signal (90 volts AC) can operate the bell. The end office sees an open circuit for DC (due to C_1) so it knows that the set is on-hook. Taking the phone off-hook moves the hookswitch to the alternate position, connecting the normally-closed dialer contact, the receiver, and the carbon microphone onto the line. The DC current from the end office battery now flows through the dialer contact, receiver, and carbon microphone. The incoming voice signal is an AC signal carried on top of the DC; this causes the receiver (SP_1) to reproduce the AC portion as sound, just as a normal loudspeaker would. The receiver essentially ignores the DC current. This phone uses pulse dialing; when the user operates the telephone dial, the dialing contact S_{2A} alternately breaks and makes the circuit according the number dialed. The end office detects the interruptions in DC loop current and begins the call routing process.

In order to transmit speech onto the line, the DC current on the line must be varied in step with the vibrations of the speech signal (in other words, the AC speech signal must be superimposed onto the DC line current). In early telephones, a *carbon microphone* was used for this purpose. It is still used in a few non-electronic telephones. A carbon microphone converts sound into a varying electrical resistance. (This is quite different than microphones normally used in communications, which convert speech into a small AC signal.) Figure 14-11 shows a cross section of a carbon microphone disc.

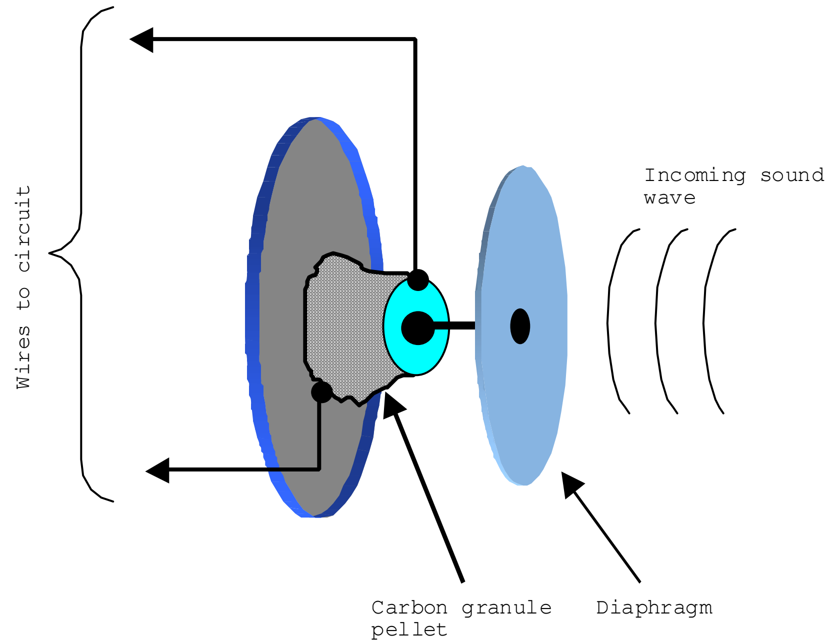

Figure 14-11: Carbon Microphone Action

In the microphone of Figure 14-11, the diaphragm collects the sound vibrations and concentrates them upon a pellet of carbon granules. This alternately compresses and releases the particles of carbon, which causes the pellet's resistance to vary in step with the information. Carbon microphones provide an immense amount of signal gain -- typically 30 to 60 dB in excess of what a conventional microphone can provide, but they require a DC bias source to operate. The early telephone system wouldn't have been possible without the carbon microphone, since electronic amplification hadn't yet been invented! Modern telephones use either dynamic or electret condensor microphones with solid-state amplification to achieve the same results without the inherent distortion caused by the carbon microphone.

Sidetone Control

The phone of Figure 14-10 will work to transmit and receive speech from a telephone line, but has one annoying problem. When the user speaks into the microphone, their voice will be echoed back quite loudly in the earpiece (SP_1), since the earpiece shares the same DC circuit as the carbon microphone. In response, most people will either speak more softly, or move the microphone away from their mouth, which will greatly reduce the speech level going to the line. Transmitted audio that is heard in the receiver is called *sidetone*. A correct level of sidetone is beneficial; it helps people to speak with the proper voice volume on the phone, and also lets them know that the line is "live."

The circuit of Figure 14-10 produces too much sidetone level. You might think that we could just put a resistor across the earpiece to reduce sidetone, and you'd be right; however, this also has the very undesirable side effect of reducing the sensitivity of the telephone receiver to incoming speech from the other party. Instead, the circuit of Figure 14-12 is used to partially cancel the transmitted speech signal.

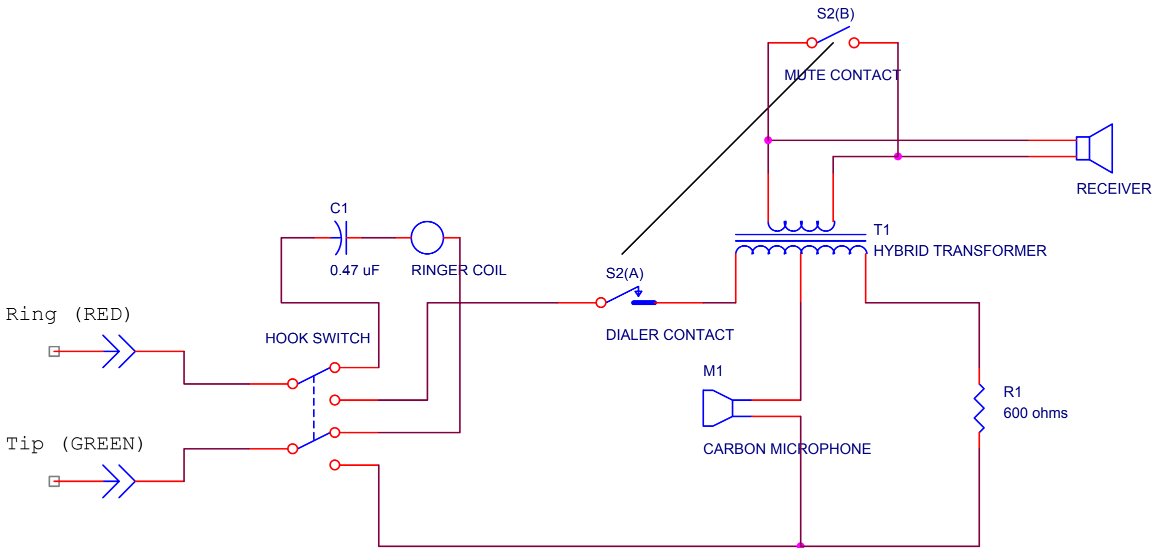

Figure 14-12: A telephone Set with Sidetone Suppression

The circuit of Figure 14-12 has been modified by the addition of the *hybrid transformer* T_1 and the *balancing resistor* R_1. T_1 has three coils with equal numbers of turns; the bottom two coils are connected in series to form the primary. Each of the coils in T_1 always has the same AC voltage drop. The operation is straightforward. Incoming voice signals form an unbalanced voltage drop across the entire primary (bottom two windings) of T_1. This causes the voice signal to be reproduced on the secondary (upper) winding of T_1 and therefore it is heard in the receiver. When the user speaks into the microphone, the DC line current is modulated as before, but in the primary of T_1 two *opposing* signals are created, since the microphone signal is driving the center tap. Therefore, only a small amount of microphone signal shows up on the secondary of T_1, canceling most of the sidetone. For this cancellation to take place, T_1 must see a balanced impedance on each side of its primary winding. To the left, the phone line provides an impedance of about 600 Ω, so resistor R_1 is set to 600 Ω to provide this balance. There's another enhancement in this circuit unrelated to sidetone. In the circuit of Figure 14-10, the dialing process interrupts the DC loop current, producing a very loud clicking noise in the receiver. Mechanical dialers provide a *muting* contact that closes whenever the dialer is moved away from its mechanical stop. The muting contact bypasses the earpiece, preventing the annoying clicks.

Electronic Telephone Circuits

The circuit of Figure 14-13 is the first step toward an all-electronic telephone unit. The mechanical pulse dialing mechanism has been replaced with a Sanyo LC7365N DTMF generator IC. The carbon microphone circuit persists; it could be replaced with a condensor microphone and amplifier circuit. This is typically the case in modern units.

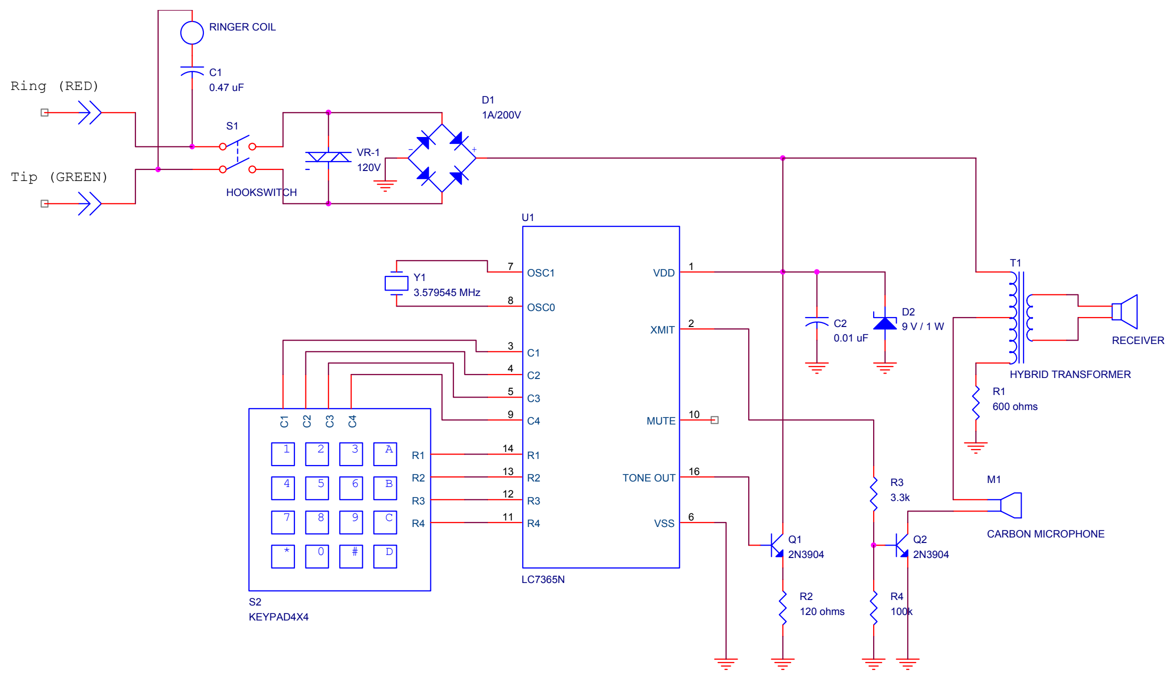

Figure 14-13: A Telephone Set with DTMF Capability

In the figure you can see that we simplified the hookswitch circuit. The ringer is now permanently wired across the line, with C_1 to block the DC component. (If this connection were made with the phone of Figure 14-12, the bell might "tinkle" during the dialing process due to inductive transients caused by making and breaking the circuit.) The hookswitch enables the electronics to be coupled to the line; VR_1 is a transient suppressor, which absorbs any high voltage spikes that might appear on the line before they can damage the sensitive solid-state circuitry. Because telephone lines in a home are often wired with the wrong polarity (which would be bad for the electronics!), bridge D_1 is used to assure that the proper line voltage polarity feeds the internal works of the phone. When the phone goes off-hook, zener diode D_2 limits the supplied voltage to 9 V to protect the DTMF generator chip, U_1.

U_1 contains all the logic needed to generate the DTMF signal. It has an internal crystal oscillator (an external TV colorburst crystal is used), plus a keyboard scanning circuit. The keypad is directly connected to the row and column inputs (C1-C4, R1-R4) of the IC. The IC automatically scans the keyboard; when a key is pressed, it drops the *XMIT* pin low (to disable the local microphone, so that local sound isn't transmitted with the DTMF tones, which could cause an error in reading them at the end office) and provides DTMF audio at the *TONE OUT* pin, which is buffered by Q_1. The DTMF audio is directly impressed across the line (power supply) by Q_1. When the key is released, the *TONE OUT* signal shuts off, and the *XMIT* pin again goes high, turning on Q_2, which again enables the local microphone. The hybrid network is identical to that in Figure 14-12, and performs exactly the same function.

Finally, note that the circuit of Figure 14-13 has been "stripped" of RF bypass capacitors (except for C_2), which would add slightly to the complexity of the circuit. A well-designed telephone should be immune to strong radio frequency fields, but unfortunately, manufacturers often omit a few pennies worth of bypass capacitors for the sake of cost cutting. Thus, many modern electronic telephones are quite sensitive to RF interference, especially from HF and VHF transmitters. In particular, the addition of a 0.01 µF / 600 V