CDC 3C251A

Communications-Computer Systems
Control Journeyman

Change Supplement for Volume(s): 1, 2, 3, 4 and S01

IMPORTANT: Make the corrections shown in this supplement before beginning your study of the volume(s) it affects. This supplement has both pen-and-ink changes and replacement pages. Tear out the replacement pages and insert them in your volumes.

Extension Course Institute
Air University
Air Education and Training Command

Changes for the Text: Volume 1

Pen-and-Ink Changes:

Page-Col	*Subject*	*Line(s)*	*Correction*
Inside front cover		First-5	Change "MSgt Timothy L. Bearden" to "MSgt Lisa Heurter" and change e-mail address to "heurter@335trs.kee.aetc.af.mil."
i	Preface	3-5 fr bot	Change e-mail address to "heuter@335trs.kee.aetc.af.mil" and change author information to "335 TRS/UOCA, ATTN: MSgt Lisa Heurter, 600 Hangar Road, Keesler AFB, MS 39534-2235."
1-69	URE-5	4-Last fr bot	At the end of each option add a "W."
2-25		Last	At the bottom of the page, add a new paragraph: "***Current flow.*** If we close the switch in figure 2-15, A, current will flow. The three ammeters would show how much current is flowing at various points in the."
2-36		4	Change "W" to "Ω."
		12	Change "W" to "Ω."
2-44		8 fr bot	Change "3" to "30."
2-47	Step 2,3,4	Equations	Change all occurrences of "W" to "Ω."
2-49	Step 3	Equation	Change all occurrences of "W" to "Ω."
2-50	Step 4	Equation	Change all occurrences of "W" to "Ω."
2-78		10 fr bot	Change "p" to "π."
2-79		17, 19	Change "p" to "π."
2-80		11	Change "frequency" to "period" and delete "referred to as."
		12	Change "frequency" to "period."
2-85			Change "2p" to "2π."

Page Changes:

Remove:	*Insert:*
2-97 – 2-100	2-96a – 2-100j
2-105 – 2-106	2-104a – 2-106
2-111 – 2-112	2-111 – 2-112

capacitor to charge with a reverse polarity. From T4 to T6, the direction of capacitor current is just the opposite of current flow from T2 to T4.

An important thing for you to see at this point is that with a sine wave of voltage applied, the capacitor has its peak voltage charge when current is zero (T2 to T4). Also, with zero voltage on the capacitor, maximum rate of change of applied voltage causes maximum capacitor current. The capacitor current is directly proportional to the rate of change of voltage across the capacitor; however, capacitor current leads capacitor voltage by 90°.

Look at the first 90° of the sine wave (T1 to T2) and try to realize that electrons must flow into one plate and away from the other plate of a capacitor before there can be a voltage across the capacitors. If you can understand this, you can understand that capacitors store energy during voltage rise and discharge energy as the applied voltage decreases. Keep in mind, though, that current does not flow through the capacitor; the charging and discharging of the two capacitor plates cause the current to flow in the circuit. Capacitors do not consume energy. They only store it. Therefore, a pure capacitor dissipates no power.

Calculating capacitive reactance. The opposition of a capacitor to the flow of current is called *capacitive reactance*, X_c. Let use see what factors determine this reactance. Refer to figure 2-73, A, which shows a capacitor in series with an alternating-voltage source. The capacitor stores an amount of energy equal to its capacity times the voltage. This amount off energy is actually stored twice (+E and –E) in each cycle of applied voltage (fig. 2-73, B). The time allowed for charging depends on the frequency $\left(f = \frac{1}{t}\right)$. If the frequency is doubled with no change in C or E, the same energy (Q) is stored in half the time. When the frequency is doubled (fig. 2-73, C), the capacitor charges to the peak voltage twice as many times, using twice the current. If the frequency is cut in half (fig. 2-73, D), one-half the current flows in the same time.

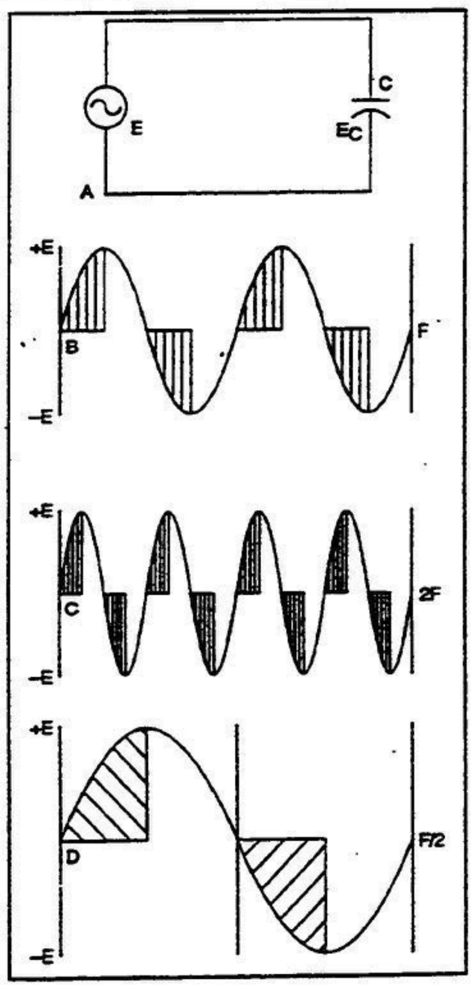

Figure 2-73. A capacitor connected in a series circuit.

Capacitive reactance, then, depends on frequency. As frequency increases, current increases, proving that the opposition (X_c) decreases.

Capacitive reactance also depends on the size of the capacitor. As capacitance increases, a larger capacitor allows more current flow to charge to the same voltage. thus, capacitive reactance decreases as the size of the capacitor increases.

We can now define capacitive reactance by the following formula:

$$X_c = \frac{1}{2\pi fC}$$

where:

X_c = capacitive reactance in ohms

2π = 6.28

f = frequency

C = capacitance in farads

Now we know there are two variables that affect capacitive reactance. They are the size of the capacitor and the frequency of the circuit. Since the formula for X_c always involves $\frac{1}{2\pi}$, which is $\frac{1}{6.28}$, we can save time later if we divide 6.28 into 1. We obtain 0.159, and the formula can now be written as:

$$X_c = \frac{.159}{fC}$$

Also, by transposing, you can quickly change the formula to determine capacitance when X_c and f are known:

$$C = \frac{.159}{fX_c}$$

or we can solve for an unknown frequency when X_c and C are known:

$$f = \frac{.159}{CX_c}$$

Notice there is a similarity in the calculation of total capacitance to the calculation of total resistance. Capacitors add in parallel. This is easy to see when you recall that capacitance is directly proportional to plate surface area. If two or more capacitors are connected in parallel, you find the equivalent capacitance by adding their values.

$$C_t = C_1 + C_2 + C_3 + \ldots$$

Capacitors in series must be treated like resistors in parallel to determine the equivalent capacitance. In effect, the thickness of the dielectric material increases and the distance between the plates combines to produce an equivalent capacitor smaller than any one of the series capacitors.

With two capacitors, use the formula:

$$C_t = \frac{C_1 \text{ x } C_2}{C_1 + C_2}$$

For any number of series-connected capacitors of equal value, the total series capacitance is equal to their total value divided by the number of capacitors (as with equal resistors in parallel):

$$C = \frac{C}{N}$$

For more than two unequal capacitors in series:

$$C_t = \frac{1}{\frac{1}{C_1} + \frac{1}{C_2} + \frac{1}{C_3}}$$

In using the formula for capacitors in series, the total capacitance is smaller than the smallest capacitor. The combination of capacitors in series results in an equivalent capacitor that has a smaller value and withstands a higher circuit voltage than any one of the series capacitors.

When the capacitive reactance (X_c) for a capacitor is known, the value is expressed in ohms. In circuits that contain capacitors in series and parallel and capacitive reactance values are given, treat the values as resistance. Remember that if the values are in ohms, treat them as you do resistors.

In a series circuit, the total capacitive reactance is expressed:

$$X_c \; total = X_{c_1} + X_{c_2} + X_{c_3} + \ldots$$

In a parallel branch, use the formulas for resistors in parallel: Three or more reactances:

$$X_{c_t} = \frac{1}{\frac{1}{X_{c_1}} + \frac{1}{X_{c_2}} + \frac{1}{X_{c_3}}}$$

Two reactances:

$$X_{c_t} = \frac{X_{c_1} X_{c_2}}{X_{c_1} + X_{c_2}}$$

Any number all the same value:

$$X_{c_t} = \frac{X_c}{N}$$

In solving for capacitor voltage, current, or reactance, use Ohm's law as follows:

$$E_c = I_c X_c$$

$$I_c = \frac{E_c}{X_c}$$

$$X_c = \frac{E_c}{I_c}$$

A good capacitor stores a charge. A capacitor that is open or shorted is useless because it will not store or hold a charge. Capacitors having values greater than 0.001 µF can be checked for opens or shorts using an ohmmeter. Before you check a capacitor with an ohmmeter, ensure that it is fully discharged and disconnected from the circuit.

2-4. Logarithmic Units of Measurement

Now that you've learned logarithms and basic electronics, you're ready for some application. This lesson will familiarize you with the systems controller's primary unit of measurement—the decibel.

032a. Figuring gain and loss in decibels

Decibel basics

If a one-millivolt signal is input to an amplifier with an amplification factor of 1,000, you would expect the output to be 1 volt (.001 X 1000). With the same amplification factor, you would expect a 2-millivolt input to produce a 2-volt output. If those 2 millivolts resulted in something other than 2 volts, though - say 1.8 volts - the device would be *nonlinear*, which means the output is not directly proportional to the input.

As systems controllers we tend to be more concerned with the way a signal sounds to the human ear than with its voltage value, and the way our ears perceive sound is nonlinear. To illustrate this, imagine yourself at the symphony. As the orchestra plays, you perceive very soft parts and very loud parts, and a wide range in between. The loudest part may actually contain 100 times more sound energy than the softest part, but it won't sound 100 times as loud to you. Our ears respond to sound logarithmically; to a human listener, a doubling of sound power doesn't make it sound twice as loud. This logarithmic response allows us to hear an extremely wide range of sound intensities.

Quite often, changes or differences in sound intensity are more significant than absolute sound values. For this reason, we have a logarithmic unit of measurement, the *bel*, which represents a *ratio between two quantities*. The bel is rather large, though, so we use the *decibel* (one tenth of a bel), abbreviated *dB*.

The decibel is a power ratio expressed as a logarithm multiplied by 10. It is useful for representing signal gain or loss, and for describing the measured power of a signal as compared to some reference power.

Figuring gain and loss

In our business we must deal with equipment that increases or decreases the power of voice signals. If the output of a device is greater than the input, we say there has been a *gain*. Amplifiers produce signal gain. If the output is less than the input, there has been a *loss*. Pads, or attenuators, introduce loss.

To figure the amount of gain or loss in bels, you would use this formula:

$$\text{bels} = \log \frac{P_1}{P_2}$$

Where P_1 is usually the output power, and P_2 is the input power.

To figure this gain or loss in decibels, you must multiply the logarithm by 10:

$$\text{decibels} = 10 \log \frac{P_1}{P_2}$$

Let's say we have an amplifier that takes a 1 milliwatt signal and amplifies it to a 5 milliwatt signal. What's the gain in dB?

$$\begin{aligned} \text{dB} &= 10 \log \frac{\text{output}}{\text{input}} \\ &= 10 \log \frac{5}{1} \\ &= 10 \log 5 \\ &= 10 \text{ X } .6990 \\ &= 6.99, \text{ or about 7 dB} \end{aligned}$$

What happens if the output of the device is lower than the input? For example, the input is 2 milliwatts and the output is 1 milliwatt? Well, let's use the formula and see:

$$\begin{aligned} \text{dB} &= 10 \log \frac{\text{output}}{\text{input}} \\ &= 10 \log \frac{1}{2} \\ &= 10 \log .5 \\ &= 10 \text{ X } \bar{1}.6990 \\ &= 10 \text{ X} -.301 \\ &= -3.01 \end{aligned}$$

When we talk about gain and loss, the plus and minus are understood; we don't say there's been a "plus 3-dB gain", or a "minus 6-dB loss." In the above example, we have a 3-dB loss. Because the plus and minus are understood, you can use an easier method to figure loss. This method, shown in the next example, will keep you from having to use negative characteristics in the logarithm.

Since you know from your input and output figures that there has been a loss, just put the larger value on top. You will get the same number for your answer, but it will be positive. Let's use the same numbers we used for our last example, and give it a try.

Our pad has an input of 2 milliwatts, and an output of 1 milliwatt. We know this represents a loss, because the output is less than the input. Normally we would divide output by input, but we know this will result in a number less than one. To avoid having to work with a negative logarithm, we can put the input value on top:

$$dB = 10 \log \frac{2}{1}$$

$$= 10 \log 2$$

$$= 10 \text{ X } .301$$

$$= 3.01 \text{ dB}$$

Now, wasn't that easier? Although this answer is positive, you must remember to state it as a loss.

A very useful fact to remember is that any time sound power is doubled, there is a gain of 3 dB. If sound power is cut in half, there is a loss of 3 dB.

So far we've discussed the decibel in terms of sound *power*, but it can also represent voltage or current ratios.

The relationship of power, voltage, and resistance is such that $P = \frac{E^2}{R}$

where:

P = power

E = voltage

R = resistance

If you substitute $\frac{E^2}{R}$ in the decibel formula, you get:

$$dB = 10 \log \frac{E_1^{\ 2} / R_1}{E_2^{\ 2} / R_2}$$

Now, if the input and output resistance are the same, the R's cancel out, and you get:

$$dB = 10 \log \frac{E_1^{\ 2}}{E_2^{\ 2}}$$

$$= 10 \log (E_1/E_2)^2$$

Now, one of the properties of logs is that the log of $a^2 = 2 \log a$. What we end up with for our formula is:

$$dB = 10 \text{ X } 2 \log (E_1/E_2)$$

$$= 20 \log (E_1/E_2)$$

This is the formula we use to calculate gain when our input and output signals are measured in volts.

To figure gain or loss in decibels when only current is known, you must remember the relationship between power, current, and resistance, which is:

$$P = I^2R$$

where:

P = power

I = current

R = resistance

Substituting this formula in the dB formula yields:

$$\begin{aligned} dB &= 10 \log \frac{I_1^{\,2}R}{I_2^{\,2}R} \\ &= 10 \log \frac{I_1^{\,2}}{I_2^{\,2}} \\ &= 10 \log (I_1/I_2)^2 \\ &= 10 \text{ X } 2 \log (I_1/I_2) \\ &= 20 \log (I_1/I_2) \end{aligned}$$

It's important to remember that these formulas using voltage and current are only good if the input and output resistance are the same. If input and output resistance are different, the formulas become:

$$dB = 20 \log \frac{E_1\sqrt{R_2}}{E_2\sqrt{R_1}} \text{ for voltage, and } dB = 20 \log \frac{I_1\sqrt{R_1}}{I_2\sqrt{R_2}} \text{ for current.}$$

032b. Referenced dB measurements

dBm

Earlier we stated that one of the uses of the decibel is to compare a measured value to a reference value. It's important to remember that the decibel itself is not an absolute quantity; it expresses ratios—comparisons between like quantities: volts to volts, watts to watts, milliwatts to milliwatts. Therefore, a ratio in dB doesn't specify what units are involved, only how much difference there is between two values. A difference of +13 dB could be 1 milliwatt to 20 milliwatts, or it could be 1 kilowatt to 20 kilowatts.

Often in our work it's convenient to express or represent *absolute* power with a logarithmic unit, and there is a way we can do this with the decibel.

If we agree on a reference power, measurements can be compared to it, and can be written as so many dB above or below it. For the dBm, the reference is 1 milliwatt of power. From this we get the definition for dBm: a unit of absolute power in dB referred to 1 milliwatt. The formula is:

$$\text{dBm} = 10 \log \frac{P}{1}$$

where:

P = measured power in milliwatts

1 = one milliwatt, the reference power

Actually the values we get from using this formula are still ratios. They are ratios between a fixed power and various measured powers. But because we do specify a reference power, the values also express absolute, or real, powers.

If a measured power equals the reference power, we have 0 dBm. If the measured power is larger than the reference power, the dBm value is positive, and if it is smaller, the dBm value is negative. Let's look at some example calculations, with the first one being the case where the measured value equals the reference value.

$$\begin{aligned} \text{dBm} &= 10 \log \frac{1 \text{ mW}}{1 \text{ mW}} \\ &= 10 \log 1 \\ &= 10 \text{ X } 0.0000 \\ &= 0 \end{aligned}$$

Suppose the measured power is double that of the reference power. We should get +3 dBm. Let's see if we do:

$$\begin{aligned} \text{dBm} &= 10 \log \frac{2 \text{ mW}}{1 \text{ mW}} \\ &= 10 \log 2 \\ &= 10 \text{ X } 0.3010 \\ &= 3.0 \end{aligned}$$

Conversely, if the measured power is half that of the reference power, the answer should be –3 dBm.

$$\begin{aligned} \text{dBm} &= 10 \log \frac{0.5 \text{ mW}}{1.0 \text{ mW}} \\ &= 10 \log 0.5 \\ &= 10 \text{ X } \bar{1}.6990 \\ &= 10 \text{ X } (-0.3010) \\ &= -3.0 \end{aligned}$$

To avoid working with negative characteristics in your logarithm, you can put the larger value on top and then assume your answer is negative. You should never have to actually make this kind of computation, though. Normally you will use a test set that will measure the signal, compare it to the 1 milliwatt reference power, and display a positive or negative dBm value.

dBm0

Signals must pass through many pieces of equipment as they travel through a typical systems control facility. Quite often, equipment items will introduce gain or loss on the signal. It's very important for you to know where these gains and losses occur, so you can take them into account when you are taking and reporting signal measurements.

Test level points or *transmission level points* (TLP) are points where there's an engineered gain or loss on a circuit. TLPs are assigned dBm values. The engineered gains and losses they represent actually change your reference for the purpose of measuring test tones and other signals. For example, at a –16 TLP, the reference is no longer 0 dBm; it has changed to –16 dBm. This change in reference could potentially cause a lot of problems, if we didn't have a unit of measurement to account for it. Fortunately, we do have such a unit—dBm0. The formula for dBm0 is:

$$\text{dBm0} = \text{dBm} - \text{TLP}$$

where:

dBm0 = measurement referred to the TLP.

dBm = actual measurement.

TLP = reference level of the test point.

Using dBm0, you can easily see gains and losses in the system, taking into account the engineered values at the various TLPs. Imagine you're at Croughton, and you're testing a circuit with a controller at Ramstein. The distant controller sends you a 0 dBm tone, and you receive the tone at a –15 dBm. Has your channel really experienced 15 dB of loss? Maybe not. You suddenly remember that you're testing at a –16 TLP. You do a quick calculation to figure your receive level in dBm0:

$$\begin{aligned}\text{dBm0} &= \text{dBm} - \text{TLP}\\ &= (-15) - (-16)\\ &= (-15) + 16\\ &= +1\end{aligned}$$

Rather than a signal loss, you have a 1-dB gain! This probably doesn't happen very often between Croughton and Ramstein, but it was just an illustration.

We systems controllers rarely use a 0-dBm tone. Most often we use the DCS standard test tone level, which is –10 dBm0. At a –16 dBm TLP, this tone would have to be –26 dBm:

dBm0 = dBm - TLP; therefore dBm = dBm0 + TLP

In this case, dBm = (–10) + (–16) = –26.

You may hear the term *dBr* used. It means the same thing as TLP. For instance, a +7 TLP means the same thing as a +7 dBr point.

The value of working with dBm0 is that everyone knows what is meant. If two controllers are checking a circuit with DCS standard test tones and the receiving controller says he's getting it at –12 dBm0, the sending controller knows the tone is arriving 2 dB lower than it should be. The sending controller doesn't have to know anything about the reference level of the receiving controller's test point.

Other references

We defined dBm as "decibels referenced to 1 milliwatt," and this is the unit you will use most often for signal power measurements, but it's important to remember the decibel can represent several different ratios. The one you use depends on your application. Don't be surprised, then, if you see the unit dBV, which means "decibels referenced to 1 volt," or dBW, which is "decibels referenced to 1 watt."

VU

If you're measuring channels with speech or music—signals that fluctuate continuously—a dB meter will try to keep up with every fluctuation of signal power and will be hard to read. VU meters are best for this type of application, because they have damping to "smooth out" the peaks and valleys of complex signals. VU meters are analog meters; they have a needle that moves up and down a scale as signal power increases or decreases. A standard VU meter is calibrated with a 1-milliwatt reference at 600Ω, so 0 VU equals 1 milliwatt of power. Although VU and dBm have the same reference power, there's not really a direct correlation between the two units, because VU meters and dB meters respond to complex signals differently. If you have a VU measurement you want to correlate to dBm, subtract 1.4 from the VU reading for an approximate dBm reading.

032c. Noise measurements

When we measure noise, we have to establish a weighting factor as well as a reference power. The reason for this is that noise measurements always apply to a certain bandwidth which must be stated or implied. Weighting refers to the amount of attenuation, if any, given each frequency within the stated or implied bandwidth.

The frequency response of a noise measuring meter should closely match the bandwidth of the channel under test. In the case of voice circuits, we are concerned only with the noise that fouls up our communications. To measure this noise properly, we developed weighting filters with bandpass characteristics such that the amount of noise allowed to pass through the filter is in proportion to the effect it has on the user. The weighting filter is inserted between the channel under test and the meter, so the meter only responds to the noise of interest. Normally the filter is physically a part of the noise measuring set.

The only basic difference between a level measuring set and noise meter is the weighting network. You can make a noise measuring meter from any dB meter by externally connecting a weighting filter.

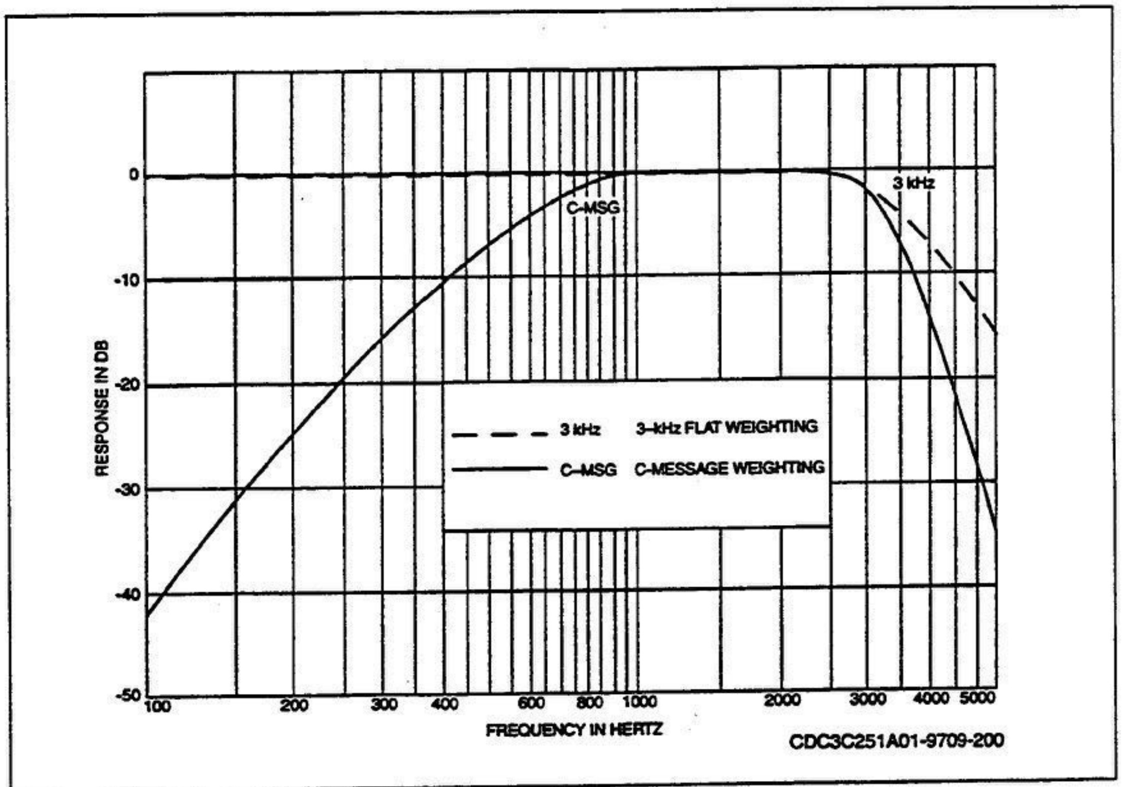

Figure 2-74. Noise weighting.

The filters you will use most often are the *3-kHz flat* and *C-message*. The 3-kHz flat filter has a "flat" response—it allows you to measure noise across the entire bandwidth of a voice channel. As you can see in figure 2-74, the 3-kHz flat filter's response drops off somewhat at the extreme high end of the band, but noise in this region is of little consequence even in voice communications. With the C-message filter, you measure noise mainly through the center of the band, because this is where noise can have the most impact, especially for data circuits. Noise measurements are assumed to be 3-kHz flat unless some other weighting is specified. The following is a short description of the noise units you'll work with most. Remember, you must account for TLPs with these units, just like with signal power measurements. A "0" after the unit indicates you've accounted for the TLP.

dBrn

This is an expression of weighted circuit noise power in dB referred to 1 picowatt (–90 dBm). The 1 picowatt of power or –90 dBm is called *referenced noise*, hence the "rn" in the abbreviation. Zero dBrn = –90 dBm, so the closer you get to 0 dBrn, the quieter the channel. On the dBrn scale, your readings will be positive; e.g. 33 dBrn always means +33 dBrn.

dBrnC

This is noise power in dBrn measured by a noise measuring set with C-message weighting. The reference noise power is 1 picowatt. With C-message weighting, 1 milliwatt of white noise, randomly distributed over a voice channel, will read about +88.5 on a noise meter. At the other end of the scale, 1 picowatt of white noise power spread over the same band will read about –1.5 dBrn on the meter, because of the frequency weighting. This is about the only time you'll ever see a negative dBrn reading.

dBm

In noise power measurements this expression refers to noise power in dB referenced to 1 milliwatt (0 dBm). Noise values in dBm are always negative, unless the noise power is more than 1 milliwatt (and if that's the case you definitely have a problem). When reading noise on a dBm scale, you want to see values that are more negative; the closer to –90 dBm, the better. Most noise standards are specified in dBrn or dBrnC rather than dBm, but you have to either measure in dBm or convert to dBm when figuring signal-to-noise ratio.

Noise power conversion

Theoretically, it's a simple matter to convert from one noise measurement unit to another. You just have to use the right conversion factor. Now, whether or not this is the right thing to do is a different story.

Conversion factors will make your answers truly theoretical because they are based on the assumption that you only measured white noise that was evenly distributed across the entire channel. The noise you measure on a real channel or system is seldom (if ever) true, flat, white noise. More often than not, there will be interfering tones or noise impulses on your channel. They could cause a big difference between what you would measure with a C-message filter and what you would measure with a 3-kHz flat filter. If you have both of these filters and your specifications call for a noise measurement in a particular unit, use that unit with the appropriate filter.

Conversion does have its place, though, mostly in planning, engineering, and developing standards. Here are three common noise measurement conversion formulas:

dBrn = dBm + 90

dBrnC = dBm + 88.5

dBrnC = dBrn –1.5

032d. Correction and error factors

In the communications field (at least here in the U.S.), most circuit test points have 600 ohms impedance. Likewise, most noise and level meters are calibrated to read voltage across a 600-ohm impedance. What happens if you need to take a measurement at a test point with something other than 600 ohms impedance? Simple — just use a correction factor. Correction factors adjust dB measurements to account for differences between a test set and test point impedance. The following formula shows you how to derive the correction factor:

$$\text{dB (correction factor)} = 10 \log \frac{Z_m}{Z_{tp}}$$

where:

Z_m = meter scale calibration impedance.

Z_{tp} = test point impedance.

Here's an example. Suppose you are taking a level measurement at a 75-ohm test point, using a meter calibrated to 600 ohms. To determine the correction factor:

$$\begin{aligned} \text{dB} &= 10 \log \frac{600}{75} \\ &= 10 \log 8 \\ &= 10 \text{ X } .9031 \\ &\cong 9 \end{aligned}$$

You can use the old "dodge" of always putting the larger value on the top of the fraction—that way your answer is always positive. If the test point impedance is lower than the impedance of the test set, add the correction factor to your reading. If the test point impedance is higher, subtract the correction factor. Using the above example, we would add our 9 dB correction factor to the reading we obtained at the 75 ohm test point. In other words, if our meter read –20 dBm, the actual signal level would be –11 dBm (–20 + 9 = –11).

Error factors

What happens if you connect your level meter to a circuit's monitor jack, intending to make a bridging measurement, but you forget to take the meter out of "term" mode? Your circuit will be double-terminated, and your meter reading will be about 3.5 dB lower than it should be. If someone plugs a meter right into the line jack of a circuit, but leaves the meter in bridging mode, the reading will be 6 dB higher than it should be. That's because the bridging resistor has such high impedance it makes the circuit act like it's open. Nearly all the applied voltage of the circuit will drop across that resistor, and the level reading will be high. If you're doing in-service QC's and all your readings are coming out 3 or 6 dB off, that's a hint that you should check your test point and your test set configuration. You *can* apply the error factors to the readings you got, but it's always better to re-take your measurements in the proper mode.

Self-Test Questions

After you complete these questions, you may check your answers at the end of the unit.

025. The basic concepts of AC

1. What rule is used to determine direction of induced current?

2. Is induced voltage directly or indirectly proportional to the strength of the field?

3. What is used in an alternator to make contact with the slip rings?

4. As a conductor is rotated to the magnetic field, when is maximum voltage induced into it?

5. What is the complete wave (360°) of induced voltage called?

026. The peak, effective, and average voltage of selected waveforms

1. Define waveform.

2. Describe four common waveform shapes.

3. Define peak voltage.

4. Define peak-to-peak voltage.

5. Define effective voltage.

6. Define average voltage.

8. Solve for X_c in the following problems:

Frequency	*Capacitor Value*
a. 1 kHz	0.5 μf
b. 400 Hz	0.01 μf
c. 20 kHz	0.05 μf

9. Solve for total capacitance:

   a. Three 0.05 μf capacitors connected in parallel.

   b. Three 0.05 μf capacitors connected in series.

   c. Three parallel capacitor legs, one leg a 0.05 μf capacitor, the other two legs are 0.1 μf capacitors each.

032a. Figuring gain and loss

1. Define the terms *gain* and *loss*.

2. What is the gain through an amplifier if the input is 5 milliwatts and the output is 10 milliwatts?

3. The input to a pad is 7 milliwatts, and the output is 1 milliwatt. What is the loss through the pad in dB?

4. You have measured the input to a device at 12 millivolts, and the output at 20 millivolts. Input and output resistance are the same. Figure the amount of gain in dB.

032b. Referenced dB measurements

1. Define dBm.

2. The power of a signal is .1 mW. Convert this reading to dBm.

3. You measure a signal at –29 dBm at a –16 TLP. What is the signal level in dBm0?

4. What type of meter is best for measuring power levels on channels carrying speech or music?

032c. Noise measurements

1. What is noise weighting?

2. When recording signal noise measurements, how do you indicate that you've taken the TLP into account?

3. What is the reference noise power for dBrn?

4. Why is it best to use the proper noise weighting filter rather than convert from one noise unit to another?

032d. Correction and error factors

1. Using a meter calibrated to 600Ω, you have measured a signal at -7 dBm0 at a 1200Ω test point. Compute the correction factor and determine the actual signal level.

2. How does double termination affect dB meter readings?

Answers to Self-Test Questions

017

1. Matter is anything that occupies space and has weight.
2. A substance that cannot be reduced to a simpler substance by chemical action.
3. A compound is the resultant substance formed from the chemical combination of two or more elements which cannot be separated by physical means.
4. A mixture is a combination of elements or compounds that are not chemically combined and can be separated by physical means.
5. The smallest particle of a compound that has all the characteristics of that element.
6. Protons, neutrons, and electrons.
7. The outermost shell of an atom.

9. The valence of an atom determines its ability to gain or lose electrons.
10. Negative ions are atoms which have more than their normal number of electrons and carry a negative charge and positive ions are atoms which have less than their normal number of electrons and carry a positive charge.
11. The process by which an atom gains or loses electrons.

018

1. One that has more or less than the normal number of electrons.
2. Like charges repel each other; unlike charges attract each other.
3. The coulomb.

019

1. Conductors, semiconductors, and insulators.
2. Conductors have many free electrons; insulators have very few free electrons.
3. The opposition to current flow.
4. Ohm; omega (Ω).
5. A conductor's resistance doubles as its length is doubled, however, doubling its cross-sectional area only increases its resistance by one-half its original value.
6. Resistivity is a term we use to describe the relative opposition that various materials offer to current flow.
7. An increase in temperature causes the resistance of metallic conductors to increase and the resistance of nonmetallic conductors to decrease.
8. When the resistance of a conductor decreases as its temperature increases.
9. The ability of a material to conduct an electrical current.
10. The unit of measurement for conductance is the "mho" and its symbol is "G."
11. A material's conductance is the reciprocal of its resistance, and vice versa.
12. Because of its high conductivity, comparatively low cost, and its physical characteristics.

020

1. The flow of electrons through a conductor.
2. The relative number of free electrons in a material.
3. Point y to point x.
4. Potential difference; volts.
5. Force that causes electrons to move through a conductor.
6. The rate at which electrons pass a given point in a conductor.
7. The symbol used to represent current is "I."
8. The unit of measurement for current flow is the "ampere", or amp, and its symbol is "A."
9. Chemical cell and mechanical generator.
10. (a) 1.5V.
    (b) 2.2V.

7. The opposition of a capacitor to the flow of current.

8. (a) 318Ω; (b) 750 kΩ; (c) 159 MΩ.

9. (a) $C_t = 0.15$ µf; (b) $C_t = 0.0167$ µf; (c) $C_t = 0.25$ µf.

032a

1. If the output of a device is greater than the input, the device is said to produce a gain. If the output is less than the input, there is a loss.

2. 3 dB.

3. 8.4 dB.

4. 4.4 dB.

032b

1. A unit of absolute power in dB referred to 1 milliwatt.

2. −10 dBm.

3. −13 dBm0.

4. VU meter.

032c

1. The amount of attenuation, if any, given each frequency within a stated or implied bandwidth.

2. By including a zero after the noise unit, e.g. dBrn0, dBrnC0.

3. 1 picowatt.

4. Because you shouldn't assume the noise measured is flat, white noise evenly distributed across the entire channel bandwidth.

032d

1. Correction factor is 3 dB. Since the test point impedance is higher than the meter impedance, we subtract the correction factor for an actual reading of -10 dBm0.

2. It lowers readings by about 3.5 dB.

Do the Unit Review Exercises (URE) before going to the next unit.

Unit Review Exercises

Note to Student: Consider all choices carefully, select the *best* answer to each question, and *circle* the corresponding letter.

31. (017) When two or more elements are chemically combined, what is the resulting substance called?
    a. An atom.
    b. A mixture.
    c. A molecule.
    d. A compound.

32. (017) Electrons in the outer shell of atoms are called
    a. photons.
    b. neutrons.
    c. valence electrons.
    d. ionized electrons.

33. (018) A body of matter that has an equal number of protons and electrons in each atom is
    a. ionized.
    b. uncharged.
    c. positively charged.
    d. negatively charged.

34. (018) Which of the following diminishes in proportion to the square of the distance from a charged body?
    a. Fields of force.
    b. Number of protons.
    c. Number of electrons.
    d. Directivity of the charges.

35. (019) What is the unit of resistance?
    a. Amp.
    b. Ohm.
    c. Volt.
    d. Henry.

Changes for the Text: Volume 2

Pen-and-Ink Changes:

Page-Col	*Subject*	*Line(s)*	*Correction*
Inside front cover		First-5	Change "MSgt Timothy L. Bearden" to "MSgt Lisa Heurter" and change e-mail address to "heurter@335trs.kee.aetc.af.mil."
i	Preface	13-15 fr bot	Change e-mail address to "heuter@335trs.kee.aetc.af.mil" and change author information to "335 TRS/UOCA, ATTN: MSgt Lisa Heurter, 600 Hangar Road, Keesler AFB, MS 39534-2235."
1-5		4 fr bot	Change "slowly" to "quickly."
B-1		2-3	Change "Howard & Sams Company" to "Howard W. Sams & Company."

Changes for the Text: Volume 3

Pen-and-Ink Changes:

Page-Col	*Subject*	*Line(s)*	*Correction*
Inside front cover		First-5	Change "MSgt Timothy L. Bearden" to "MSgt Lisa Heurter" and change e-mail address to "heurter@335trs.kee.aetc.af.mil."
i	Preface	13-15 fr bot	Change e-mail address to "heuter@335trs.kee.aetc.af.mil" and change author information to "335 TRS/UOCA, ATTN: MSgt Lisa Heurter, 600 Hangar Road, Keesler AFB, MS 39534-2235."
1-35	409	13-14	Change to: "**409. Characteristics of bi-phase shift keying (BPSK), quadrature-phase shift keying (QPSK), and di-phase shift keying (DPSK).**"
1-59		First	Change "Mc(t)" to "M_c(t)."
		8	Change "m1(t)" to "m^1(t)."
1-66	409		Add the following answers: "7. By changes in phase value from one sample to the next. 8. It requires a less complex receiver."
3-21		13 fr bot	Between "192" and "384" insert a space.
3-31		12	Change "99" to "98."
	Fig 3-18	Legend	Change "99" to "98."

Page Changes:

Remove:	*Insert:*
1-39 – 1-40	1-38a – 1-40
2-39 – 2-44	2-39 – 2-44
3-43 – 3-44	3-43 – 3-44e
3-47 – 3-50	3-47 – 3-50a

DPSK

Differential phase shift keying (DPSK), also called di-phase shift keying, is a modified form of phase shift keying. In DPSK, the phase shift is not to an absolute phase value but simply to the "other" of two possible phases. For example, in one DPSK scheme, the phase of a carrier is shifted only when a 0 is transmitted. If, at one sampling moment, there is no phase change from the preceding sample, a 1 is being transmitted; if there is a change, a 0 is being transmitted. Thus information is not represented by an absolute phase value but by a *change* in phase value. The main advantage of DPSK is that it requires a less complex receiver than a basic PSK signal. A demodulator need only detect changes in phase, not absolute phase values. The presence of the carrier is not required for detection.

Self-Test Questions

After you complete these questions, you may check your answers at the end of the unit.

408. Significant phase modulation characteristics

1. In phase modulation, how is the transmitted signal modulated?

2. What effect does a change in amplitude have on a PM signal?

3. What effect does a change in frequency have on a PM signal?

4. What are the characteristics of the sidebands of a PM signal?

5. How is carrier stability maintained in a PM signal?

409. Characteristics of bi-phase shift keying (BPSK), quadrature-phase shift keying (QPSK), and di-phase shift keying (DPSK)

1. In BPSK, what amount of phase shift of the carrier is used to represent binary 1's and 0's?

2. What are the advantages of bi-phase modulation?

3. What are the disadvantages of BPSK?

4. What are the main advantages of QPSK over BPSK?

5. How is phase ambiguity overcome when phase modulation techniques are used?

6. How many carrier phases does quadriphase modulation have?

7. How is information represented in DPSK?

8. What is the main advantage of DPSK?

1–5. Pulse and Delta Modulation

The modulation schemes covered in the preceding sections all used a sine-wave carrier. The mathematical treatment of them assumed sine-wave modulation for simplicity. Actual modulating signals are more complex, as we said. Occasionally, pulse trains or rectangular waves are used to modulate the carrier. Radar, for example, uses a train of rectangular pulses to amplitude modulate a sine-wave carrier. This is conventional AM or DSBEC and *not* a form of pulse modulation, since the carrier is a sine wave.

As another example, teletype pulses are sometimes used to frequency-modulate an audio-frequency sine-wave carrier to put the data signals on a voice communications channel. The resultant signal is called *frequency shift keying (FSK);* it is not pulse modulation, since the carrier is a sine wave. FSK can be thought of as digital FM. Finally, *phase shift keying (PSK)* is frequently used to interface digital pulses to a voice communications system. This is one of the rate applications of PM and also is not pulse modulation, because the carrier is a sine wave. The pulse-modulation techniques to which we devote the remainder of this text are modulation techniques in which the carrier is a pulse train or a rectangular wave.

410. Types of pulse modulation and their operational characteristics

Consider the rectangular wave shown in figure 1–43. The wave can be described or characterized by its peak voltage, E_c; its period, T_P; and its pulse width or pulse duration, T_D. Each of these parameters can be varied by an information signal to produce a modulated rectangular pulse train.

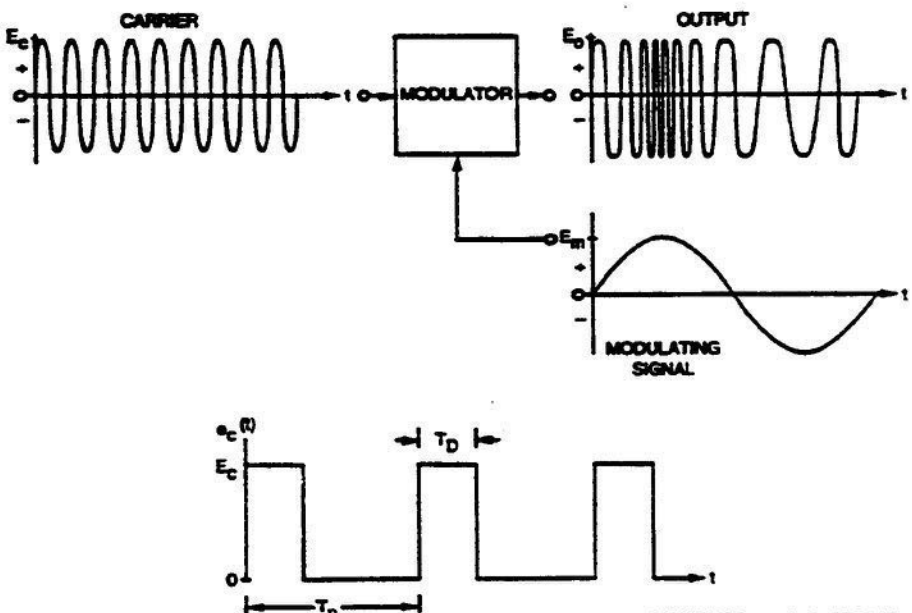

Figure 1–43. Rectangular pulse-train time frequency representation.

modulation required in TDM telephone equipment is not needed. The process is one of simply interweaving telegraph channels into a composite TDM signal that is compatible with the particular transmission subsystem.

In its simplest form, the operation of telegraph TDM equipment is similar to that of parallel-to-serial and serial-to-parallel conveters. That is, when the equipment is sending, parallel inputs are converted to a single serial output. Conversely, when the equipment is receiving, a single serial input is converted to a number of parallel outputs.

The modulation rate of the single serial stream depends on the number of associated telegraph loops and the modulation rate on the loops. For example, if 16 unit interval signals of 75 baud are time-division multiplexed, the modulation rate of the serial stream would be 16 x 75 = 1200 baud. Typical telegraph TDM equipment used in the DCS can handle up to sixteen 60-, 75-, or 100-word-per-minute DC telegraph loops.

419. System characteristics of FDM and TDM

A brief comparison of FDM and TDM is presented in the following text. We are making the comparison with regard to their primary communication requirements concerning bandwidth and noise.

Bandwidth requirements. In single-sideband FDM subsystems, the bandwidth of the composite FDM signal is equal to the number of channels times the bandwidth of each single channel. For example, in a 12-channel system with a 4-kHz bandwidth per channel, the composite signal has a bandwidth of 48 kHz. In a double-sideband FDM system, this bandwidth is doubled; i.e., 96 kHz. These bandwidths are relatively narrow when they are compared with those required in practical TDM subsystems. In a 64-step PCM/TDM system handling twelve 4-kHz channels, the composite TDM signal would have frequency components exceeding 576 kHz, or 48 kHz per channel. This is 12 times greater than the bandwidth required for single-sideband FDM and 6 times greater than the bandwidth required for double-sideband FDM.

If the available transmission subsystem bandwidth is restricted because of technical or economic reasons, FDM is the proper choice. This would be true where the available bandwidth of the transmission subsystem is derived at substantial cost, such as the present-day tropospheric scatter radio and submarine cable systems. If the available bandwidth is not restricted, TDM may be a better choice, since it provides better performance with regard to overall circuit noise and is easier to encrypt.

Noise. From a performance standpoint, the noise in FDM systems increases as the system's length is increased. Noise is cumulative because as it is introduced into the system, it adds to and is amplified with the signal at all repeater stations and terminals. As the system's length is increased, more repeaters and terminals are required, resulting in high overall circuit noise between the originating and terminating user terminals.

In TDM systems, the elements (bits) in the composite TDM signal are usually regenerated at each repeater station or terminal. As we use it here, "regeneration" refers to the process of generating a "clean" pulse upon the receipt of a "noisy" pulse.

Noise in TDM systems stays relatively constant between terminals regardless of distance. This is true as long as the noise in the transmission subsystem links does not exceed the threshold of recognition for a proper regeneration of the TDM pulse.

419a. Time slot interchange and time-space-time division switching

Time slot interchange

In a time-division multiplexed bit stream, data occurs during time slots in a frame format. In order to establish a communications channel through a switch, this data must be switched from its incoming time slot to a time slot on an outgoing bit stream. *Time slot interchange (TSI)* is the process of interchanging time slots from one bit stream to another. TSI is used in the Digital Patch and Access System (DPAS), Integrated Digital Network Exchange (IDNX), and in digital telephone switches.

Time slot interchange is different from time division multiplexing, but the two are used together in a digital switching system. The purpose of TDM is to allow multiple incoming circuits to use a single outgoing line. A TSI unit is sort of like a rail yard, where train cars are switched from one track to another. True, a train is normally a lot slower than a digital data stream, but maybe the analogy is useful. Let's say the 5:05 from New York gets into the Chicago station on track 2. That same train must then become the 5:35 to St. Louis, and must depart on a different track. In a TSI unit, data comes in on a particular channel on a particular data stream, and is switched to a different channel on a different data stream. The switching isn't done with gears or levers, though. There's not even a physical connection made between the two channels. The very sophisticated hardware and software of the TSI unit just make sure the frames representing those two channels are in the same place at the same *time*.

In systems like DPAS and DISN, time slot interchange allows connections to be made at channel level between incoming and outgoing trunks. This gives systems controllers great flexibility in routing and testing individual circuits.

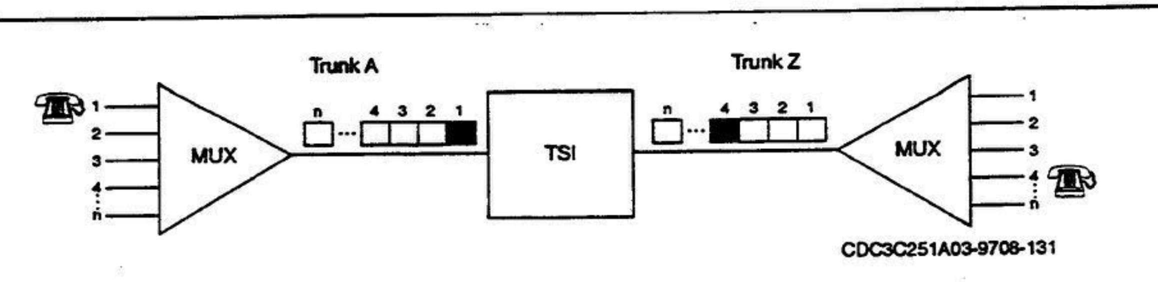

Figure 2–17. TSI macro view.

Inputs to a typical TSI system may be digitized voice or data. The usual format is PCM. Each incoming PCM frame is assigned a time slot in a serial data stream for input to the time slot interchanger. You can see a macro view of the TSI process in figure 2–17. Keep in mind this is a greatly simplified view, just to show you the basic concept. In the figure, a voice circuit originates on channel 1 of a TDM. The output of that mux is a data stream we'll call Trunk A. Our voice circuit is in time slot 1 of that data stream. The TSI switches that circuit to time slot 4 on outgoing Trunk Z. Trunk Z is the input to another TDM. Time slot 4 on Trunk Z becomes outgoing channel 4 and the connection is made.

What really makes TSI work is memory. Refer to figure 2–18. Incoming frames are written sequentially into a buffer or *speech memory* for temporary storage. Time slot 1 is assigned to memory word 1, time slot 2 goes to memory word 2, etc. Another memory, the time-slot memory or *control memory*, stores the information about where to transfer the incoming time slots. Data is read out of the buffer randomly, as the appropriate output time slots become available. For example, if incoming time slot 3 has to go to time slot 1 on the outgoing bit stream, it may be read out of the buffer memory before incoming time slot 1, which has to wait for outgoing time slot *n*.

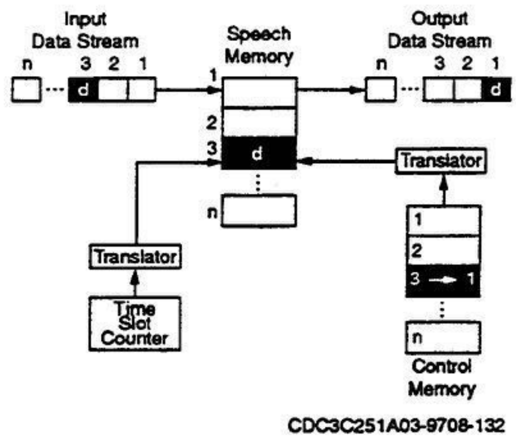

Figure 2–18. TSI operation.

You may have guessed that this process requires very precise timing. The memory must be reconfigured for each incoming frame. The time slot counter synchronizes the incoming and outgoing data by controlling the transfer of data in and out of memory. In figure 2–18, input data frame *d* is in time slot 3. When the time slot counter counts "3," the translator writes *d* to location 3 in both the speech memory and the control memory. The control memory knows that the data in input slot 3 needs to go to output slot 1. When the time slot counter counts "1," the output translator reads *d* out of the buffer memory in outgoing time slot 1.

For a full-duplex circuit, the TSI process must make connections in both directions—if time slot 2 is connected to time slot 12 on the transmit side, then time slot 12 must be connected to time slot 2 on the receive side.

It's interesting to note that the incoming and outgoing channels are completely independent of each other. They are connected together, but rather than by physical cross-connect, they are connected by *time*. Also, the time slot interchange matrix doesn't establish permanent paths for a circuit or call; it is reconfigured for every incoming frame. It happens so fast, though, that it's completely transparent to us; from our vantage point the channel appears to have physical continuity.

Time-space-time division switching

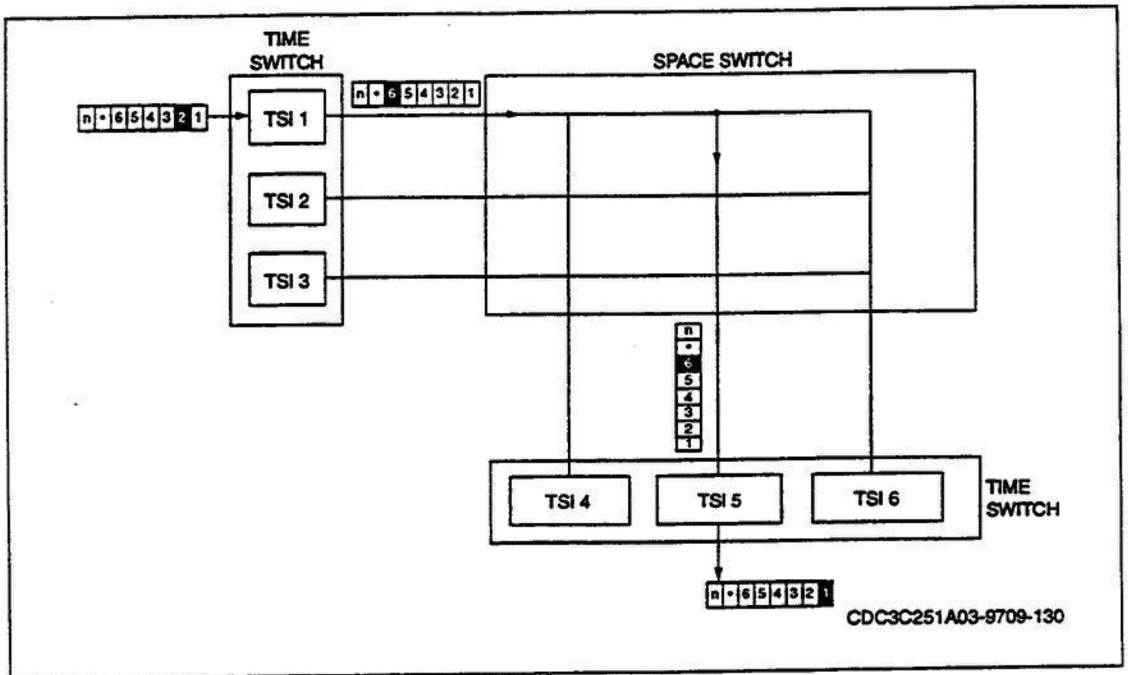

Figure 2–19. Typical TST switch.

Time slot interchange is an element of *time division switching*, where circuits are connected to each other by time. In *space division switching*, circuits are physically connected, either through mechanical relays or logic matrices. In large switching systems it becomes necessary to use combinations of switching techniques, to accommodate a large number of possible connections. A common configuration is a *time-space-time (TST) division* switch. A typical TST configuration is shown in figure 2–19. The space division part of this switch is a matrix of logic gates. Control signals turn the gates on and off during each time slot to route data from one TSI to another, but it's important to note that no time slot interchanging takes place in this matrix; it's a physical connection, so the data is simply routed through. In the example shown, data comes into TSI 1 in time slot 2 and is output from TSI 5 in time slot 1.

Self-Test Questions

After you complete these questions, you may check your answers at the end of the unit.

417. TDM sampling, bandwidth, and synchronization

1. Is the time duration of a TDM sample critical? Why?

2. How is the TDM sampling rate determined?

3. What sampling rate is commonly used in TDM systems?

4. What effect does an increase in the number of channels in a TDM system have on its bandwidth?

5. What are the bandwidth requirements for a TDM system?

6. How can bandwidth requirements for TDM systems be reduced?

7. How is signal synchronization maintained in the TDM process?

8. What constitutes a TDM "frame"?

418. Solving problems concerning frame period or sample rate of TDM subsystems

1. What type of data stream is used in TDM?

2. What must be done to analog signals before they can be added to a TDM bit stream?

3. Briefly describe the voice multiplex technique used in TDM.

4. What would be the frame period of a VF channel when the highest frequency in the channel is 4kHz? How do you arrive at this frame period?

5. What would be the pulse-modulated sample period for a 12-channel, VF multiplexer with a frame period of 125 microseconds. How is the result derived?

419. System characteristics of FDM and TDM

1. What system, FDM or TDM, would be preferred if you have a limited bandwidth with which to work? Why?

2. What system, FDM or TDM, provides the least amount of protection against noise interference? Why?

3. What system, FDM or TDM, offers the best overall system noise performance? Why?

419a. Time slot interchange and time-space-time division switching

1. What is the difference between time slot interchange and time division multiplexing?

2. How is time slot interchange beneficial in systems like DPAS and DISN?

3. Briefly explain the basic TSI process.

4. Briefly explain how a combination of time- and space-division switching is used in a TST division switch.

Answers to Self-Test Questions

412

1. Low frequencies are associated with speech intensity.
2. High frequencies are associated with speech intelligibility.
3. Speech signal levels are normally maintained between 0 dBm0 and –31 dBm0.

4. Requirements for speech transmission are less stringent than data transmission because up to 75 percent of the information content in speech is redundant.
5. Voice frequency (VF) band; 300 to 3400 Hz.
6. It masks or interferes with the signal.
7. Thermal agitation, radiation, and mismatched antenna and waveguide components are some causes of noise.

413

1. The frequency range of a lower sideeband, 12-channel group is 60 to 108 kHz.
2. 84 to 88 kHz.

3. 102.996, 90.996, 78.996, 70.996, and 62.996 kHz.
4. Guard bands prevent channel interaction.
5. Channel filter quality determines the total usable frequency bandpass for each channel.
6. Guard bands also provide unused portions of the frequency spectrum that can be used to insert signaling or pilot tones.
7. Supergroups have a 60-channel or 5-group capacity.
8. 525 kHz (612 kHz - 87 = 525).
9. Supergroup 2 is not modulated with a supergroup carrier.
10. One master group has the capacity of 10 supergroups, 50 groups (5 groups per supergroup), and 600 channels (12 channels per group).
11.

Channel	*Group*	*Supergroup 1*	*Master Group*
2	3	413 kHz	199 kHz
5	1	329 kHz	283 kHz
8	5	533 kHz	79 kHz
10	2	397 kHz	215 kHz
12	4	501 kHz	111 kHz

12. TLPs provide known reference points for measuring system performance against established parameters.
13. a. 0–4 kHz; 80–84 kHz; 480–484 kHz; 480–484 kHz.

    b. 2600 Hz; 69.4 kHz; 398.6 kHz; 213.4 kHz.

    c. 1004 Hz; 94.996 kHz; 517.004 kHz; 1094.996 kHz.
14. a. –27.5 dBm0.

    b. Example 2: –36 dBm; –56 dBm; –38 dBm; –65 dBm.

    Example 3: –26 dBm; –46 dBm; –28 dBm; –55 dBm.

414

1. (1) a, (2) b, (3) e, (4) c, (5) f, (6) g, (7) d, (8) h, (9) i.
2. To provide for alignment as a signal level reference, give alarm indications for alarm circuitry, provide for master frequency oscillators synchronization, and to be used for automatic gain control (AGC) level regulation.
3. Group pilot; synchronizing pilot.
4. Group pilots are usually inserted along with the VF channel frequencies of each group, while the synchronizing pilot is inserted at the line frequency band.
5. Group pilots go through the same frequency translation and amplification as the channels, so, what happens to one happens to the other.
6. The level of the pilot has varied above or below the nominal level.
7. The synchronizing pilot generated by the "master" FDM terminal is used to control the frequency of the "slaved" terminal's master frequency oscillator.

415

1. Information signaling is an alert that announces incoming calls. Supervisory signaling is the conveying of information regarding switch hook conditions. Control signaling is the conveying of dialing or digital information to establish the desired connections.
2. Out-of-band is a signaling frequency within the guard band between VF channels; in-band is a signaling frequency within the VF channel frequency band.
3. Out-of-band signaling circuits have to be built into the multiplex channel equipment, sharper cutoff frequencies are required of channel filters, and DC repeaters are required at the end of each link. In-band signaling tones are within the channel speech band, which could lead to talkdown.
4. Common channel signaling is a signaling method using a link, common to a number of channels, necessary for the control, accounting, and management of traffic on these channels.
5. It eliminates some of the special equipment for signaling and the channel filters associated with this equipment.
6. (1) j, (2) b, (3) c, (4) d, (5) a, (6) f, (7) o, (8) h, (9) i, (10) l, (11) k, (12) g, (13) m, (14) n, (15) e.

416

1. a. Passes frequencies below the cutoff frequency.

   b. Passes frequencies above the cutoff frequency.

   c. Passes a band of frequencies between a low cutoff frequency and a high cutoff frequency.

   d. Eliminates a band of frequencies between a low cutoff frequency and a high cutoff frequency.
2. It is a symmetrical resistive pad used in unbalanced circuits.
3. It is a symmetrical resistive pad used in balanced circuits.
4. (1) They are used to reduce the strength of : voice-band signals before modulation, sidebands before modulation, and incoming sidebands before demodulation. (2) They are placed at the input of fixed-gain amplifiers to obtain overall gain variations. (3) They are placed at the input of amplifiers to prevent amplifier overloading. (4) They can be used in conjunction with a voltmeter to measure the gain of an amplifier.
5. The basic use of hybrids is to convert from a four-wire circuit to a two-wire circuit or vice versa. They are also used to provide impedance matching between certain circuits and isolation between other circuits.
6. Transformer.
7. Resistive.
8. Resistive.
9. Transformer.

417

1. No, because as long as the sampling rate of the highest frequency in the intelligence signal is adequate, the size and frequency of the original waveform can be regenerated.
2. By the Nyquist sampling rate definition, the sampling rate must be equal to twice the signal bandwidth which would permit almost complete regeneration of the original signal waveform.
3. 8 kHz is the most commonly used sampling rate for TDM systems.
4. Increasing the number of channels in a TDM system increases the bandwidth required to transmit the composite signal.

5. A TDM system must be able to handle the fundamental frequency and all the upper and lower harmonics produced during the modulating process.
6. TDM bandwidth requirements can be reduced by employing pulse-coding techniques and filtering the transmit signal.
7. By use of synchronization or marker pulses in the bit stream.
8. A frame is one complete set of pulses, made up of a marker pulse and one pulse from each channel.

418

1. TDM uses a serial-type data bit stream.
2. Analog signals must be converted to digital signals prior to adding them to a TDM bit stream.
3. The VF channels are divided in time and the information in each channel is transmitted during a different instant of time, but overlapping in a common frequency spectrum.
4. $t_f = \frac{1}{4}$ kHz = $1/(4 \times 10^3)$ = 250 microseconds.
5. T_s = 125/12 = 10.4 microseconds.

419

1. FDM; FDM only requires a total system bandwidth equal to the frequencies of all the channels.
2. FDM; FDM actually adds noise (thermal, atmospheric, etc.) cumulatively to the signal as it passes through each station to its final terminating station.
3. TDM; Noise in TDM systems remains relatively constant because of regeneration, which generates a clear pulse upon receipt of a noise pulse.

419a

1. TDM combines signals so they can share a single line, while TSI switches information from one data stream to another.
2. TSI allows connections to be made at channel level, giving systems controllers the ability to route and test individual circuits.
3. PCM frames enter the TSI unit and are written to memory. The information about where they are to be routed is also written to memory. When the proper time slot on the outgoing data stream comes up, the frame is read out of memory into that slot.
4. Multiple TSI units are connected to a space division switch, which physically routes data from one TSI unit to another; no time slots are interchanged in the space switch. This type of configuration is used in large switching systems.

Unit Review Exercises

Note to Student: Consider all choices carefully, select the *best* answer to each question, and *circle* the corresponding letter. When you have completed all unit review exercises, transfer your answers to ECI Form 34, Field Scoring Answer Sheet.

Do not return your answer sheet to ECI.

29. (412) In a voice channel, intensity is contained in the
   a. carrier.
   b. low frequencies.
   c. middle frequencies.
   d. high frequencies.

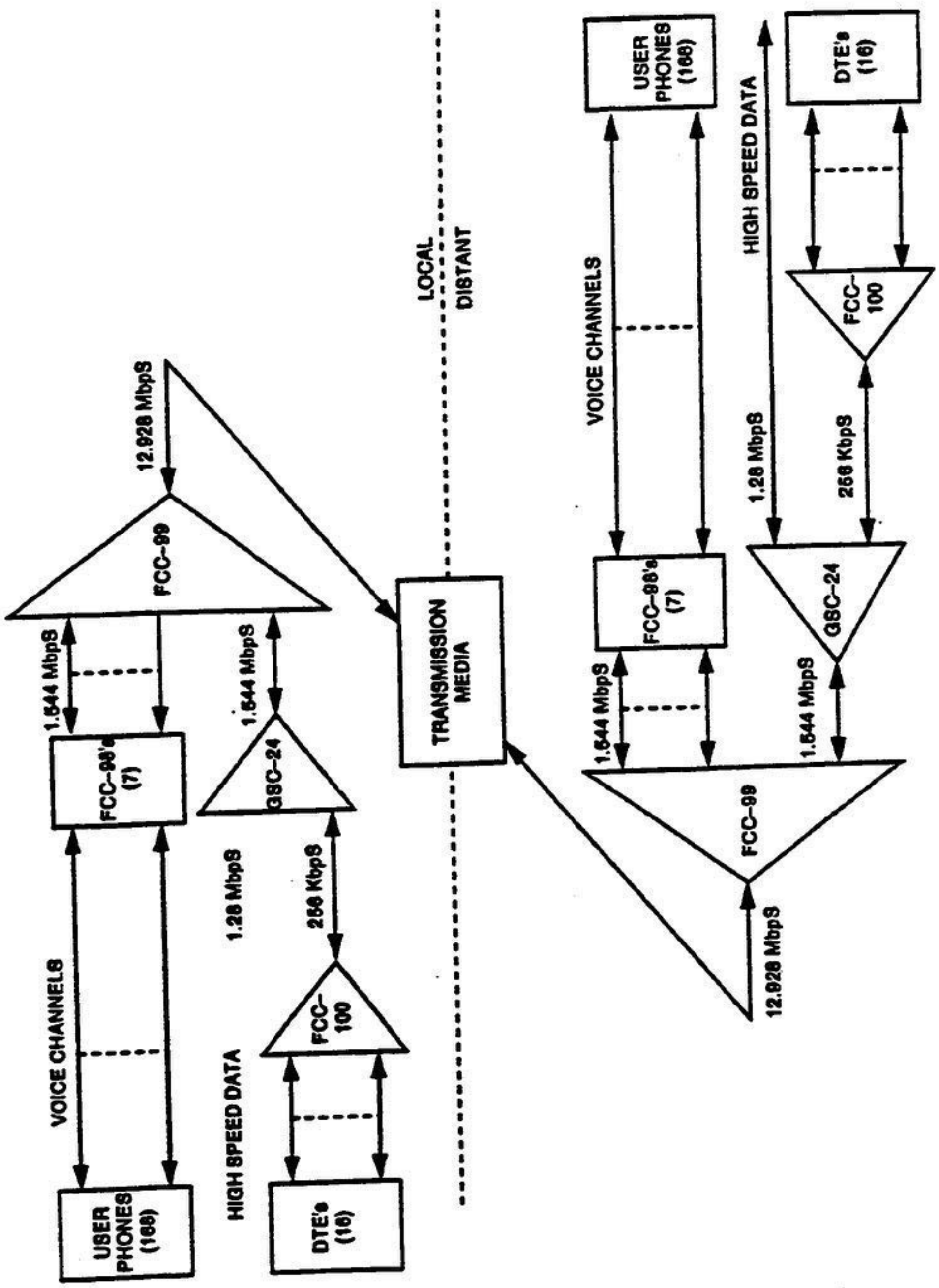

Figure 3–28. DRAMA simplified block diagram.

The multiplexer and demultiplexer modules continuously provide a smooth multiplexed and demultiplexed output stream while continuously monitoring port generator interrupt flags. The system clock for the multiplexer and demultiplexer modules is the MBS clock with each cycle being one MBS clock period. Two cycles are dedicated to moving data from the ports to the data RAM while the other two are dedicated to moving data from the RAM to the mission bit stream.

This concludes our discussion on modulation and multiplexing. We began with achieving a basic understanding of the various analog and digital techniques such as amplitude, frequency, phase and pulse modulation. Next, we applied this knowledge to comprehend the theory of multiplexing in its various forms (FDM and TDM). Finally, we completed our discussion by looking at some of multiplexing schemes employing these techniques and examples of equipment items used in the various multiplex stages. You need a good understanding of all you've learned here because there's a lot more to be covered on these subjects in the second half of the 5-level CDC.

430. Integrated Digital Network Exchange (IDNX)

The IDNX is part of the backbone of the Defense Information Systems Network (DISN). It multiplexes, demultiplexes, and switches voice, data, and video communications using the time-slot interchange process we discussed earlier in this volume. Its design is modular. There is a suite of "common equipment," and then we tack on different modules for various types of circuits. The IDNX is made by Network Equipment Technologies, Inc. There are different models available, but the one we will focus on in this lesson is the IDNX/90.

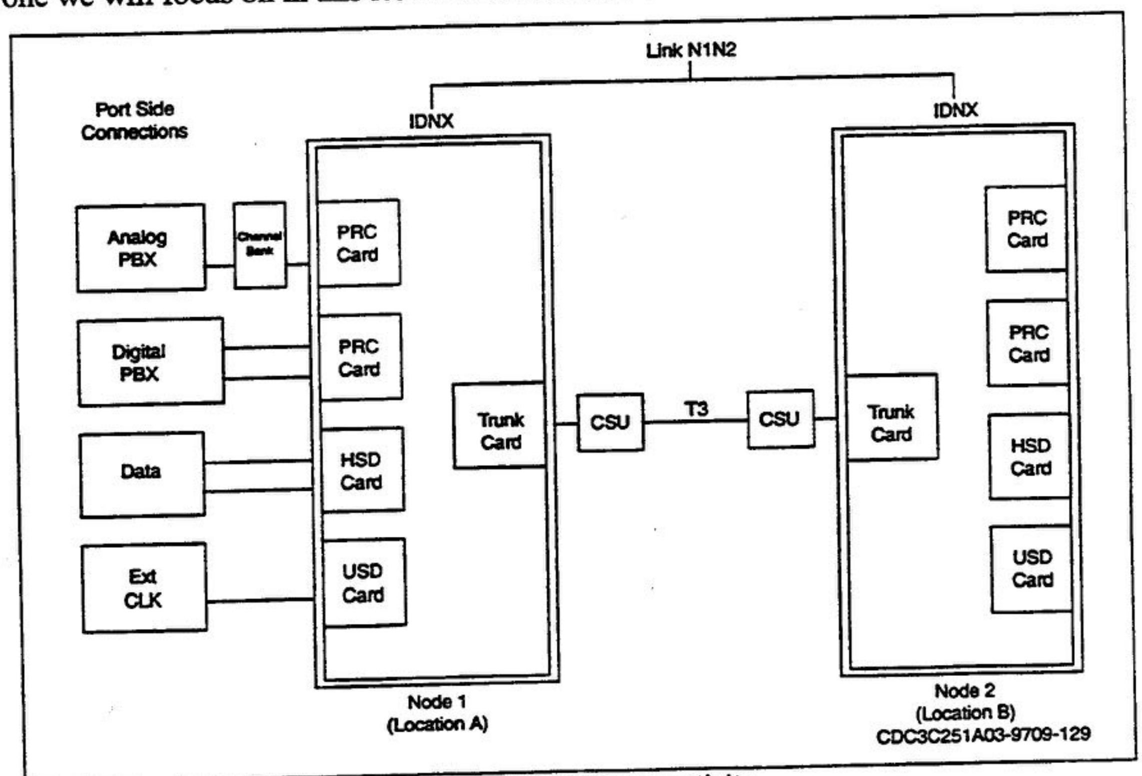

Figure 3-29. IDNX connectivity.

IDNXs can be connected to each other via leased or governm[illegible]-owned lines, using cable, microwave, or satellite media. They are normally connected to at least two other nodes. Together, the equipment and lines form a network that is highly reliable, flexible, and survivable. An IDNX *trunk* is a single, physical connection between two nodes. A *link* is a logical connection, and can consist of one or more trunks. The block diagram in Figure 3–29 shows some typical IDNX port side connections, as well as the link and trunk connecting two nodes together.

IDNX features

Dynamic bandwidth allocation

With conventional bandwidth allocation, a 9.6 Kbps circuit may be assigned to a 64 Kbps channel, because that's the smallest channel available. The remaining bandwidth on that 64 Kbps channel is wasted. Refer to Figure 3–30. The IDNX can do what's called "dynamic bandwidth allocation." It treats the trunks connected to it like a bandwidth pool. Bandwidth is allocated as needed, and when it's no longer needed it's returned to the pool for use by other circuits.

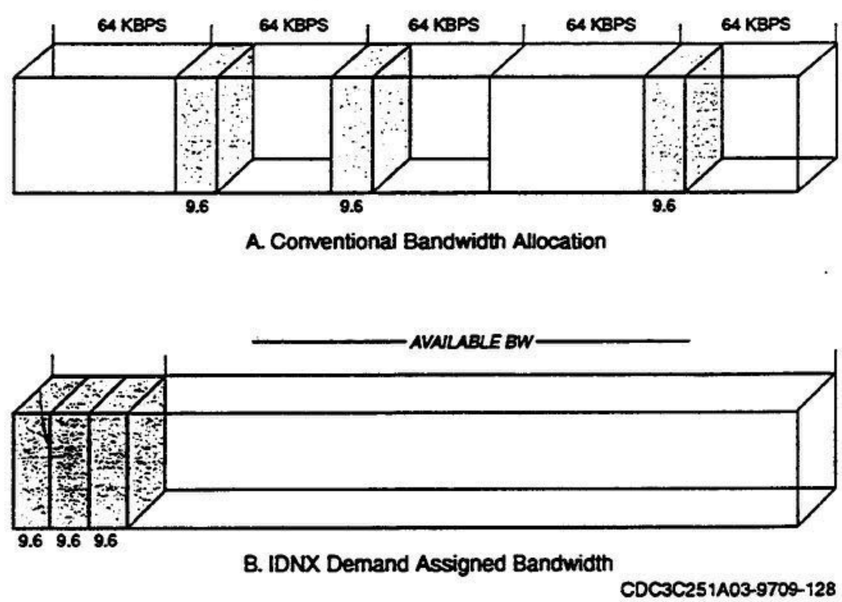

Figure 3-30. Dynamic bandwidth allocation.

Dynamic routing

The IDNX is programmed with routing tables, and can perform dynamic routing. If a path to a connected node is down, traffic can be rerouted over other trunks. This rerouting process is automatic; users may never even know there's a problem.

Transparent to users

The DISN is transparent to its users, the way your household utilities are to you. When you need electricity, you call the electric company and they provide you service; you don't need to know what cables and transformers provide your house with power. The only time you have to contact the electric company after they turn on your power is when you pay your bill or report an outage. With DISN, users request service from Point A to Point B and the DISN engineers set it up. Unless users specify certain routing (or avoidance routing), they're not concerned with the exact circuit path—as long as it works.

Centralized management

IDNXs can be centrally managed. Large networks like the DISN could become chaotic if each node site were independent of the others. Everyone would do things differently, and no one would talk to each other. With central management, the Defense Information Systems Agency (DISA) ensures there are some standards of operation throughout the network. They see the "big picture" and can isolate network problems faster than individuals at node sites. DISA can also track and analyze network trends, and can predict the effect of changes on other parts of the network.

IDNX components

Because the IDNX is designed to handle a wide variety of signal types, it has a wide variety of modules to interface these signals. There are four basic types of IDNX modules.

Common equipment modules

The common equipment modules control the inner workings of the IDNX. They don't directly connect to any external sources or devices. Common equipment modules include the switching module, processor, and memory.

Trunk modules

Trunk modules interface the IDNX with interswitch trunks to other nodes. The IDNX/90 uses only T3 or E3 (34.368 Mbps) modules. Each T3 module interfaces a DS–3 (44.736 Mbps) trunk, and an IDNX/90 can handle 4 of these trunks. That's a lot of data!

Data modules

Data modules interface digital data in synchronous or asynchronous format, at a wide range of data rates. There are eight types of data modules.

QSD (Quad Synchronous Data) and QSD–2. The QSD module has RS–232 DCE interfaces for 4 independent synchronous circuits. It supports data rates from 1.2 Kbps to 19.2 Kbps. The QSD–2 also supports 4 synchronous circuits, but has a wider range of interfaces and data rates. It supports RS–232 DCE or DTE ports from 1.2 Kbps to 19.2 Kbps; RS–530 DCE from 1.2 Kbps to 56 Kbps; V.35 DCE or DTE from 1.2 Kbps to 56 Kbps; and V.54 modem loops if the port is configured properly.

QASD (Quad Asynchronous/Synchronous Data). The QASD can do everything the QSD–2 can, and more. It supports asynchronous data at rates from 75 bps to 19.2 Kbps, and synchronous data up to 64 Kbps.

HSD (High-Speed Synchronous Data) and HSD–2. The HSD module supports two independent synchronous circuits at data rates from 9.6 Kbps to 1.344 Mbps. It provides V.35 DCE/DTE, RS–449/422 DCE/DTE, and RS–232 DCE/DTE interfaces.

The HSD–2 has an RS–530 DCE interface and supports terminal timing for DCE applications.

USD (Universal Synchronous Data). The USD is a dual-port high-speed data module that supports data rates from 1.2 Kbps to 1.344 Mbps. It supports some extra timing and clock options, and provides the following interfaces: RS–232 DCE/DTE, V.35 DCE/DTE, RS–449/422 DCE/DTE, Bi-Phase Mil–188 DCE, and RS–530 DCE.

DS0A. The DS0A module can support four independent synchronous circuits, each of which operates at a subrate of 64 Kbps (specifically 2.4, 4.8, 9.6, 16, 24, 32, 56, or 64 Kbps). Byte repetition is used to fill any unused portion of the 64 Kbps bandwidth. DS0A operation is usually considered to be pretty inefficient because user data must sometimes be repeated several times to use up the whole 64 Kbps. It can come in handy, though, in applications where data reception is highly critical, or when a transmission media is not reliable enough. The DS0A module supports RS–232 DCE/DTE, RS–232/V.54 DTE, V.35 DCE/DTE, and RD–530 DCE interfaces.

DMD (Digital Multidrop Data). The DMD is used in multipoint/multidrop applications, like weather networks. Each DMD module can support four separate circuits with RS–232 DCE/DTE, RS–232/V.54 DTE, V.35 DCE/DTE, and RS–530 DCE interfaces.

QXP (Quad X.21 Port). This module provides four separate synchronous data ports with X.21 DCE or DTE interfaces, at data rates from 1.2 Kbps to 64 Kbps.

DS0B Server Module. The DS0B module multiplexes low-speed (2.4, 4.8, and 9.6 Kbps) data circuits into a single DS0 channel. It can multiplex up to 10 subrate ports, depending on the combination of data rates used. The DS0B module has internal ports only; it doesn't interface with outside lines, but with other IDNX modules. For example, data signals from QSD, QASD, or DMD cards enter the DS0B module and are multiplexed into a single 64 Kbps channel. That DS0 signal can then be a 64 Kbps input to a PRC card (described below under voice modules).

Voice modules

Voice modules are used for analog or quasi-analog voice circuits. They can accept narrowband (3 kHz) voice circuits and convert them to 64 Kbps PCM, or they can accept PCM trunks from a PBX or channel bank. There are four types of voice modules.

PRC (Primary Rate Card). The PRC accepts two DS–1 signals in D3/D4 frame format, and separates them into 48 separate 64-Kbps channels. Each channel can then be compressed and switched individually. The PRC also supports D3/D4 Super Framing or Extended Super Framing.

TMCP (Two-Megabit Channelized Port). The TMCP has all the capabilities of the PRC, plus it can operate at the 2.048 Mbps rate commonly used in Europe. It's also compatible with European channel banks that use 30 channels instead of 24.

QAVP (Quad Analog Voice Port). The QAVP supports four analog channels that connect to PBXs, channel banks, or central offices. It provides signaling and echo cancellation, and converts the narrowband analog signals to 64-Kbps PCM data so they can be sent to their destination ports.

Voice server modules provide compression of PCM signals. Compression allows more channels to be packed into a smaller bandwidth. Server modules appear between voice port modules and trunk modules. For example, a PCM signal from a channel bank may interface with a Primary Rate Card (PRC). The PRC breaks the signal out to individual channels and sends the channels to a server module, such as the High Density Voice Compression (HDVC) card. The HDVC card compresses the data, which then goes to the trunk module for transmission.

Yes, but how does it work?

Communication within in an IDNX node takes place along *buses*. The definition of a bus is "one or more conductors that serve as a common connection for one or more devices." The buses that carry data within an IDNX shelf are built-in. Data travels along bus cables to go from one shelf to another. Buses carry different types of information, such as data, control, signaling, and clock.

When data enters the IDNX on a local port, it goes into a time slot on a bus. From there it can be routed to another local port or to a channel on an interswitch trunk, but it has to go through the switching module first. In the switching module (SX module), a time-space processor switches the data to the appropriate shelf or trunk based on instructions from the system processor.

IDNXs can be configured locally through an operator console connected to the processor module. Configuration can also be done remotely from the Regional Control Center, using a Network Management System and Sun Workstation.

Network Timing

One or more nodes in the network can be designated as "master" for timing purposes. These nodes provide clock reference signals to other nodes via trunks. Master nodes are usually those that have the most links to other nodes in the network, or have access to the best external timing sources.

The Switching Exchange (SX) module, which is one of the common equipment modules I mentioned earlier, provides clock to all other modules in the IDNX. The SX module can generate the timing signals itself (internal timing), or it can lock onto the incoming signal from a trunk module, PRC module, or USD module (external timing).

An IDNX/90 can accept up to 8 clock references, numbered 0 through 7 on the SX module. Each reference is assigned a priority that indicates which source is to be used when the highest priority clock source fails. Clock sources are set up during node initialization, and can be changed any time.

Self-Test Questions

After you complete these questions, you may check your answers at the end of the unit.

424. Equipment configurations and capabilities

1. How many PCM channels can a microwave radio that uses a 7-MHz bandwidth transmit?

2. What is the purpose of the first-level multiplex?

3. What is the purpose of the second-level multiplexer?

4. At what multiplex level is a bulk encryption performed?

425. Terms and characteristics of the AN/FCC-100

1. What type of multiplexer is the AN/FCC-100?

2. What is the maximum operating speed of the AN/FCC-100?

3. What type of data signals can the AN/FCC-100 accept?

6. What is the main difference in the framing format between the 12-channel mode and the 24-channel mode?

7. How many frames make up a superframe in a 24-channel mode?

8. What is the output rate of a 6-channel AN/FCC–98 system? Show how this is derived.

428. Characteristics and functions of the second-level multiplexer

1. What type of data stream is input to the second-level multiplexer?

2. What is the purpose of the bit stuffing operation?

3. When the T1 data streams are sampled in the second-level multiplex, how is the TDM signal divided?

4. What is another name for a T1 channel?

5. How is a second-level multiplex frame organized?

6. Describe the interface step of second-level multiplexing.

429. Characteristics of the AN/FCC–99

1. What is the function of the AN/FCC–99?

2. What is the transmit output data of the AN/FCC–99 called?

3. What is the function of the demultiplexer section of the AN/FCC-99?

4. How is data from each port kept from being mixed in memory?

5. What does the frame structure diagram illustrate?

6. What is the largest frame structure?

7. What determines the number of frame pairs contained within a superframe?

8. How many frame pairs are in a superframe at a data rate of 9.696 Mb/s?

9. Identify the different types of port modules used in the AN/FCC-99.

430. Integrated Digital Network Exchange (IDNX)

1. What is the difference between an IDNX *link* and a *trunk*?

2. Briefly describe dynamic bandwidth allocation.

3. What is the disadvantage in using DS0A configuration?

4. What is the benefit provided by voice compression?

5. How do control signals get from one IDNX shelf to another?

6. Briefly describe internal and external timing for the IDNX.

Answers to Self-Test Questions

420

1. a. To compensate for envelope delay.

   b. To multiplex each channel and combine the 12 channels to form a group.

   c. To insert a group pilot and provide 135-ohm impedance.

   d. To combine five groups into a composite signal.

   e. To modulate the output of the group multiplexer shelf to its assigned output frequency.

   f. To combine the outputs of the supergroup modulator trays.

   g. To separate the incoming line frequency into 10 supergroups.

   h. To translate one of the supergroup line frequency bands to the basic supergroup band.

   i. To break down the basic supergroup into five groups.

   j. To monitor the group pilot and provide isolation and impedance matching.

   k. To demultiplex the group signal into 12 channels.

421

1. Synchronous (STDM) and asynchronous (ATDM)/statistical (Stat-Mux).
2. Statistical multiplexing periodically redefines the length of its frames to change the number of time slots, thereby changing its number of channels. STDM permanently assigns time slots for its channels.
3. STDM transmits dummy characters on inactive time slots where Stat-Mux assigns time slots only for active channels.
4. Stat-Muxes contain concentrators with extra buffers to temporarily store data.
5. Synchronous.
6. By buffering, pulse stuffing, and interleaving.
7. By sampling the input signal at a high enough rate.
8. The formatting of TDM pulse streams into frames.
9. A longer framing pattern because of its ability to maintain synchronization without giving a false pattern.
10. The longer the framing pattern, the more the information rate is decreased.
11. Asynchronous, synchronous, and isochronous.
12. Timing differences arising from (1) slight variation in operating rates at the two ends of the circuit, and (2) variations in propagation delay over the circuit.
13. Timing.
14. Master clocks, synchronization pulses, and pulse stuffing.
15. Pulse stuffing.

422

1. To be able to take a systems oriented approach to troubleshooting.
2. To interleave numbers of channels from preceding stages into higher data rates.
3. A T1 line.
4. By channel banks designated by the levels they bridge. D1 for a T1 line (1.544-Mb/s data stream), D2 for a T2 line (6.312-Mb/s data stream), or D3 for a T3 line (44.746-Mb/s data stream).

423

1. All share compatibility with the Bell System T1 at the first-level TDM.
2. The second-level multiplexer can accommodate up to 192 channels.
3. A microwave radio set accomplishes the third-level multiplexing.

424

1. 192 PCM voice channels.
2. Multiplex up to 24 voice channels into a single T1 data stream.
3. Multiplex up to 8 data streams into a single high-speed data stream.
4. During the first-, second-, or third-level multiplexing stage or at all three stages.

425

1. LSTDM.
2. The maximum operating speed is up to 256 kb/s.
3. The AN/FCC–100 can accept synchronous, asynchronous, and isochronous data signals.
4. (1) f, (2) c, (3) a, (4) g, (5) b, (6) d, (7) e.

426

1. (1) c, (2) d, (3) b, (4) a.

427

1. Pulse-code modulation.
2. Sampling, quantizing, and encoding.
3. The least significant bit is time-shared between voice and signaling. Every sixth frame carries the signaling information.
4. 24 channels x 8-bit code word plus 1 bit for synchronization—(24 x 8) + 1 = 193.
5. 1.544 Mb/s.
6. The 12 channel mode does not use a framing bit sync (193rd bit).
7. 12 frames.
8. R_o = 6 channels x 8000 sampling rate x 8 bits/sample = 384 kb/s.

428

1. The T1 signal.
2. Stuffing bits are added to ensure that the data stream contains exactly 1,544,935 b/s to prevent the elastic storage from becoming depleted, which would include an error into the bit stream.
3. The combined signals are divided into words; a word contains one bit from each T1 channel.
4. A port.
5. A frame consists of 16 words plus a framing bit (BF) and a control bit (BC).
6. The system uses three-level detection to limit the transmission bandwidth to one-fourth the basic bit rate.

429

1. To provide second-level TDM.
2. Mission Bit Stream (MBS).
3. To demultiplex received MBS data and write it into a RAM.
4. The superframe word identifies the beginning of each new superframe and provides synchronization for the demultiplexer frame counter.
5. Frame word for frame identification, data, and possible stuff code and stuff data, as applicable.
6. Superframe.
7. The MBS rate.
8. 12 frames.
9. Bipolar, or NRZ.

430

1. A trunk is a single, physical connection between two nodes, while a link is a logical connection, and may consist of more than one trunk.
2. Channels are assigned only the amount of bandwidth they need. When the bandwidth is no longer needed, it's returned to the pool for use by other circuits.
3. The 64 Kbps bandwidth is not used very efficiently when a circuit slower than 64 Kbps is assigned to the DS0A module. On slower circuits, the user's data is simply repeated to fill up the entire 64 Kbps channel.
4. Voice compression allows more channels to be packed into a smaller bandwidth.
5. They travel along bus cables.
6. With internal timing, the SX module generates clock signals for the node; with external timing, the SX module derives a clock signal from incoming data on a trunk, PRC, or USD module.

Do the Unit Review Exercises (URE) before going to the next unit.

Changes for the Text: S01 Supplementary Material for Volume 1-4

Pen-and-Ink Changes:

Page-Col	*Subject*	*Line(s)*	*Correction*
G-3	**Carrier Frequency**	Last	After "In" add "frequency modulation, the carrier frequency is also referred to as the "center frequency." (401)"
G-15		4-5 fr bot	Change "**Peak Voltage** . . . (026)" to "**PBX**—Private Branch Exchange. A private telecommunications exchange that usually includes access to the public switched network. (430)"
G-23		11 fr bot	After "**PAM**" add new line: "**PBX** Private Branch Exchange."

AFSC 3C251
3C251A 00 S03 9802
Edit Code 01