Practical Communication Theory
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About the Author

You know perfectly well that authors write these “about the author” sections themselves, but write in the third person to somehow imply that an impartial third party is heaping on all that praise. As you will see while reading this book and then using it in your work, I have a great deal more respect for your intelligence and your valuable time than that.

The most important thing you should know about me is that I have actually used all of the working tools presented in this book, out in the “hard cruel world” where making a large mistake usually costs your boss or your customer a lot of trouble and/or money. In a few cases, it has literally been a matter of life and death. I have also spent years as a technical manager of young, eager engineers who were (and are) a lot smarter than I, but some of whom occasionally tried to defy the laws of physics in paper designs that were not always corrected before we built something that didn’t do what our customers had a right to expect (and which I enjoyed helping repair during many long nights).

With over 50 years in the business, I have participated in the design, manufacture, testing and field support of a wide range of systems and subsystems which either transmit or receive radio signals in environments, from submarines to space, and over frequency ranges from just above DC to just above light.

I hold a BSEE from Arizona State University and an MSEE from the University of Santa Clara, both with majors in communication theory. I have published 15 technical books and more than 200 articles on communications related topics in a variety of technical magazines and journals, including extensive tutorial sections in various handbooks. I have never published an integral sign (except once in a cartoon), all of these books, articles and tutorial sections have explained communications phenomena using the same practical, application oriented approach used throughout this book.

For the last 30 years, I have made my living as owner of a small company that performs communications related design studies for large companies and the government. I also teach short courses all over the world on a variety of subjects for which a practical understanding of radio propagation is essential. In fact, the real reason for this book is that so many people in those classes found the techniques presented uniquely useful to them.

If you have a technical question, need a referral to an expert in a related field, or just want to brag about a clever calculation technique to an appreciative colleague,
I would enjoy hearing from you. My email address is dave@lynxpub.com. One never knows how long a book will be around, and I will move to a consulting practice by a fishing stream in the hills one of these years, but I intend to keep the same email address as long as I can whack the computer keys so you can contact me by that fishing stream, wherever it may be.
In the 20 years since the publication of the first edition there have been no changes to the laws of physics, so it is still accurate. However, some additional, related material—especially in Chapters 4, 5, and 6—will make this second edition relevant in more situations and applications and will reflect some changes in the state of the art in components.

An antenna and propagation slide rule included with the book will allow you to make rapid calculations of antenna parameter trade-offs and radio propagation loss calculations. Section 4.4 gives detailed instructions on how to use the antenna scales on the slide rule.

Chapter 5 covered only line of sight propagation in the first edition. Now it also deals with propagation near the ground and use of the slide rule for all types of propagation calculations. Chapter 6 was limited to sensitivity calculations in the first edition but now also covers the calculation of dynamic range.

The appendixes at the end of the book have been modified to reflect the changes in the chapters.

Dave Adamy
2014
Preface to the First Edition

The key to understanding many important fields—wireless communication, television broadcasting, radar, remote control, data links, and electronic warfare to name just a few—is a practical understanding of radio propagation. You probably don’t care how radio waves propagate, but before you can specify or design any type of equipment that generates or receives radio signals—or determine whether or not it will dependably do its job in a new situation—you will need to be able to predict what a transmitted signal will look like when it gets to a receiver as a function of frequency, distance, weather, and interference. This book will provide you with practical working tools to do just that without getting into the intimidating mathematical expressions with which these considerations are explained in most communication theory texts. There are no integral signs in this book.

This is a true story. As a junior systems engineer, I took a problem to the office of an internationally recognized communications expert. I don’t remember the problem, but I distinctly remember the answer. He started with Maxwell’s equations and filled the blackboard with small print twice (elapsed time 40 minutes) before answering: “That’s what I thought, it’s about half a dB.” During the last half hour of the derivation process, I became more and more convinced that I didn’t really want to know that badly. There just had to be an easier way to solve those “half a dB” problems that come up every day.

I soon learned about a wealth of practical rules of thumb, charts and nomographs with which senior systems engineers went directly to the answers, without all the stimulating math. During the next intervening 30 years, I have collected these gems in a small, tattered notebook that has accompanied me through much of the world (never checked in luggage).

This book is a direct descendent of that precious tattered notebook—expanded with explanations—checked by mathematical derivation when I was not completely confident of the accuracy or the limitations of a rule of thumb and corrected in a few (very few) cases. For completeness, it also includes some general descriptive information about communication links, but even this is interspersed with those rules of thumb and handy charts.

After publication of the first edition, the search for handy formulas and nomographs continued. This time, they were stuck or stuffed into a tattered copy of the first edition. This second edition is expanded to include that newly discovered material.

I stand in awe of people who can look at an equation with multiple integral signs and see which way to change a variable to make a real-life system work properly.
I am not one of them. Only after I understand what is going on in a physical sense can I see and apply what the equation is trying to tell me. Therefore, devices and phenomena are described in physical rather than mathematical terms to avoid confusing you (and maybe me).

The explanations, formulas, charts, and nomographs in this book are in general accurate to 1 dB—which I have found over the years to be fully adequate for almost all system design and analysis tasks. Sometimes, of course, more accurate answers are required, involving blackboards dense with integral signs or computers cranking out 10-digit numbers. But in these cases, I have always used the approaches presented in this book to be sure that the answer was approximately right.

When a detailed analysis gives an answer grossly different from my quick one, I ask for an explanation of the assumptions. Almost always, the deeply mathematical examination is trying to break the laws of physics in some subtle way. When we fix the assumptions and then solve the problem we thought we were solving, the precise answers are consistent with the 1 dB answers. It is much better to fix that kind of problem before you freeze the design and spend a lot of money manufacturing something that is very hard to fix later.

Another very practical issue is explaining communication theory concepts in terms that the boss or the customer (who maybe used to be an engineer) will find understandable and useful. The equations in the form presented in this book and the graphical explanations should be very useful to you in that task.

You may already know much of the material presented in this book. However, in accordance with the principle that it is far better to repeat something you already know (and risk insulting the intelligence of some readers) than to leave something out (and have someone seriously confused), this book errs on the side of putting it in. Besides, most of us who have been out of school for a few years don’t mind being reminded—and we remember (with little kindness) those text authors who were overly fond of the phrase, “This is left as an exercise for the student.”

A special antenna and propagation slide rule is also provided. This handy tool will allow you to make antenna parameter trade-offs and propagation loss calculations without touching a calculator or remembering a single formula.

I hope you find the working tools in this book as useful as I have and that they keep you out of as much trouble as they have me.

Dave Adamy
Acknowledgements

It is an honor to acknowledge the contributions of the following individuals who took the time to review the draft of the first edition to this book, both for technical accuracy and for appropriateness to the audience for whom it is intended. Each gave freely of his considerable expertise and insight, and each made suggestions that significantly improved the book.

- Dr. Timothy Healy
- Professor Frederic Levien
- Mr. Michael Licata
- Mr. William Shellenberger
- Mr. Gregory Wannamaker
- Mr. Charles Weisman
- Dr. Edward Wischmeyer

It is also appropriate to recognize the many anonymous individuals who have developed the nomographs and rules of thumb enjoyed by a whole generation of communications professionals. This kind of sharing with colleagues is what makes a profession a profession.
Chapter 1
Introduction

Like most people, you probably don’t care about the mechanics of radio propagation. What everyone cares about is the quality of received signals that we perceive as the fidelity of a voice signal, the clarity of a television image, the accuracy of a radar, or the dependability of a remote control command (among others). But adequate received signal quality is achieved only when all of the elements of the communication link are proper for range and interference environment in which they must operate.

This book will allow you to select link components that will deliver the desired received signal quality, to determine the signal quality that will be achieved with a particular link or to determine the conditions (i.e., range and interference) for which a selected link will deliver adequate signal quality.

1.1 The Communication Link

The basic element of radio communication is a communication link like the one shown in Figure 1-1. It includes a transmitter, a receiver, transmitting and receiving antennas, and everything that happens to the radio signal as it passes between the two antennas.

The range of radio communication applications goes far beyond the simple case of one transmitter “talking” to one receiver. Some transmissions are broadcasts, intended for receipt by many listeners at greatly varying ranges. Some are received by unintended receivers in addition to the intended receiver, either interference or as deliberate interception. In some cases, transmissions are designed to be reflected to a receiver collocated with the transmitter. Some are designed to deliberately interfere with the reception of intended signals. However, every

Figure 1-1 Generic communication link
transmission from a transmitter to a receiver obeys the same laws of physics, so each of these cases can be analyzed—one transmission path at a time—using the same rules and formulas.

The focus of this book is on what happens to an information signal as it passes through the link. The link components are discussed functionally in terms of what each does to the signals passing through it rather than how they are implemented. Likewise, the various things that happen to signals as they propagate between the two antennas are discussed in terms of the net differences between the transmitted and received signals, with only as much discussion of the processes as necessary to allow you to understand what is happening.

Even if you are specifying or supplying only one of the link elements, you need to understand the whole link to do the job properly. Consider the broadcast situation shown in Figure 1-2. If you are providing only the transmitter, you will need to transmit adequate power to allow the farthest intended receiver to output your information signal with adequate quality, so you need to analyze the requirements of that link, making reasonable assumptions about the nature of the receiver to be used. If you are providing a receiver to receive that transmission, it must be able to perform adequately from the minimum design range to the maximum design range and to reject any interfering signals expected to be present, so you need to analyze the links from the transmitter to your receiver at minimum and maximum range. You also need to analyze the link from a worst case (or perhaps typical) interfering transmitter to determine the interference rejection specifications your receiver must meet.

Now consider the unintended receiver situation of Figure 1-3. If you are providing the unintended receiver (e.g., a monitoring receiver to be used by a government regulatory agency), you will need to analyze the link from the transmitter to the intended receiver to determine what parameters the transmitter must have to do its job. Then, you will need to analyze the link from the transmitter to your unintended receiver to determine the required specifications for the receiver you employ. The unintended link can be quite different from the intended link if, for example, the transmitter uses a directional antenna pointed at its intended receiver.
or if your receiver is much closer or farther from the transmitter than the intended receiver.

1.2 A Few Important Definitions

Some definitions commonly used in the field and used throughout this book are as follows:

- Information signal: Any type of input to a transmitter that is intended for reproduction at the output of a distant receiver. Examples are human voice, music, the video output of a television camera, a digital signal from a computer, and a radar pulse. The whole purpose of communication is to deliver this signal.
- Information bandwidth: The amount of frequency spectrum required for the information signal to retain its proper function. Examples are the amount of spectrum required for a human voice to be heard with adequate quality or the amount of spectrum required for a television signal to produce an adequate picture on the screen of a standard commercial television set.
- Modulation bandwidth: The highest modulating frequency. This is normally equal to the information bandwidth.
- Transmission bandwidth: The frequency spectrum occupied by the signal output by the transmitter.
- Required predetection bandwidth: The spectrum that must be received for a receiver to do its job. This is often (but not always) equal to the transmission bandwidth. Sometimes a receiver does not have to accept the full transmitted bandwidth of a signal to do its job. There are also circumstances in which the receiver must be wider than the transmission bandwidth of the received signal to do its job.
- Radio frequency (RF): The frequency of a transmitted signal. This is commonly used to distinguish the actual transmission frequency from intermediate frequency (IF) or audio frequency.
- Ether (or ether waves): This is an archaic term identifying the substance thought to pervade all of space. It is starting to reenter radio propagation to include any atmosphere (the earth or any other planet) and the vacuum of space.
1.3 Assumptions

The top-level assumptions made in this book are the following:

- Unless stated otherwise, the designers of equipment knew what they were doing. That is, receivers have the optimum bandwidth, transmitters use the most efficient techniques, processors use optimum algorithms, and antennas have the normally expected efficiency.
- We will deal with two propagation models: (1) line of sight propagation, wherein the transmitting and receiving antennas see each other over an unobstructed straight line and are not too close to the ground or water; and (2) two-ray propagation, wherein the transmitter and receiver are fairly close to the ground or water.
- We will be using many expressions involving the logarithm (abbreviated log) of a number. There are other types of logs (e.g., natural log, log to the base 2), but in this book log always means log to base 10 (which is what you get when you punch the LOG key on your scientific calculator).
- Unless otherwise stated, noise is white, Gaussian noise—that is, with its amplitude probability normally distributed about zero and its power density evenly distributed over the whole bandwidth considered.

1.4 Scope of the Book

The main subjects covered in this book are as follows:

- Decibel (dB) forms of values and equations (Chapter 2)
- Signal-to-noise ratio: Its definition and how it applies to communication theory (Chapter 3)
- Communication link: The characteristics and role of each part of the link (Chapter 4)
- Link equation: Predicts the performance of communication links in good weather (Chapter 5)
- Receiver system sensitivity and dynamic range: Calculation of the sensitivity and dynamic range achieved by a wide range of types of receiving systems (Chapter 6)
- Challenging propagation conditions: Characterizes and predicts link performance in bad weather, when there is not quite line of sight or when either the transmitter or receiver is moving (Chapter 7)

Each of these subjects will be covered to adequate depth to allow you to use the techniques taught to predict the performance of a radio link.

Appendixes

Derivations are generally left out of the body of the book to preserve the flow. Where appropriate, the derivations are presented in Appendixes A and B.
Four quick-formula appendixes contain all of the formulas and nomographs in Chapters 4–7. In the chapters, all equations and nomographs are explained in detail and are illustrated with examples. Once you understand how to use them, you’ll save much page-flipping time by getting just the formula or nomograph you need from the appropriate appendix.

Finally, there is an appendix with detailed instructions on how to use the included slide rule to make antenna and propagation calculations.

For consistency, metric units are used throughout the chapters of this book. However, for your convenience in dealing with requirements that define distance in statute miles or nautical miles and antenna diameter antenna height or obstruction height in feet, the quick-formula appendixes also contain formulas and nomographs in those units.
In communication theory, we spend a lot of time manipulating widely varying signal strength values. We also deal with noninteger powers and roots of numbers. The use of decibel (dB) forms of numbers and equations greatly simplifies dealing with both of these considerations.

Any number expressed in dB is logarithmic, which makes it convenient to compare values that may differ by many orders of magnitude. (Note that numbers in non-dB form are called linear to differentiate them from the logarithmic dB numbers.) Numbers in dB form also have the great charm of being easy to manipulate:

- To multiply linear numbers, you add their logarithms.
- To divide linear numbers, you subtract their logarithms.
- To raise a linear number to the $n$-th power, you multiply its logarithm by $n$.
- To take the $n$-th root of a linear number, you divide its logarithm by $n$.

To take maximum advantage of this convenience, it is common to put numbers into dB form as early in the process as possible—and to convert them back to linear forms as late as possible (if at all). In many cases, the most commonly used forms of answers remain in dB, so we can avoid converting to linear forms altogether.

It is important to understand that any value expressed in dB units must be a ratio (which has been converted to logarithmic form). Common examples are as follows:

- Amplifier gain (i.e., the ratio of the output signal strength to the input signal strength)
- Antenna gain (also treated like an amplification ratio but with some qualifiers)
- Losses (i.e., signal attenuation ratio) when passing through (a) cables; (b) switches (the off position of course has much more attenuation than the on position, but the on position still has some loss); (c) power dividers (i.e., ratio of signal power at each output port to input power); and (d) filters

To create useful equations in dB form, it is necessary to express absolute values as dB numbers. Signal strength in units of dBm is the most common example.

* If you already speak fluent dB, please feel free to skip most of this chapter. Just check out the tables and the bookkeeping diagram methodology, because we’ll be using them later.
Since dB values must always be ratios, a trick is required: Calculate the ratio of the desired absolute value to some fixed value and then convert that ratio to dB form. For example, signal strength in dBm is the dB form of the ratio of that signal strength to one milliwatt. (More on that later.)

### 2.1 Conversion to dB Form

In the following discussion, we use logarithms, abbreviated log. In this case, we are using the logarithm to the base 10, which is what you get when you punch the “LOG” key on a calculator.

The basic formula for conversion into dB is

\[
\text{Ratio (in dB)} = 10 \log(\text{Linear Ratio})
\]

For example, 2 (the ratio of 2 to 1) converts to dB form as

\[
10 \log(2) = 3 \text{ dB}
\]

(It’s actually 3.0103 dB, but everyone rounds it to 3 dB.)

And 1/2 (i.e., 0.5) becomes

\[
10 \log(0.5) = -3 \text{ dB}
\]

To make the conversion the easy way, use a scientific calculator, which has log and 10^x functions.

To convert into dB
- Enter the linear ratio (e.g., 2);
- Press the log key;
- Multiply by 10; and
- Read the answer in dB (3.0103, which rounds to 3).

To convert back from dB form to linear form, the formula is

\[
\text{Linear Ratio} = 10^{(\text{ratio in dB}/10)}
\]

For example:

\[
10^{(3/10)} = 10^{0.3} = 2
\]

To convert dB values back to linear form
- Enter the dB value (e.g., 3);
- Divide by 10 (Then hit “=” to get the value onto the display.);
- Press the “10^x” key (On many calculators this is the second function on the log key.);
- Read the answer as the linear ratio (1.99526, which rounds to 2).
2.2 Absolute Values in dB Form

As stated already, the most common example of an absolute value expressed in dB form is signal strength in dBm. This is the ratio of the signal power to 1 milliwatt converted to dB form exactly as shown in Section 2.1.

Note: dBm is a particularly important unit because many important formulas in the heart of this book either start or end (or both) with dBm values of signal strength.

For example, converting 4 watts to dBm:

\[ 4 \text{ watts} = 4,000 \text{ milliwatts} \]
\[ 10 \log(4000) = 36 \text{ dBm} \]

and, of course,

\[ 10^{36/10} = 10^{3.6} = 4,000 \text{ (milliwatts)} = 4 \text{ watts} \]

2.3 dB Forms of Equations

These equations use absolute numbers (usually in dBm) and ratios (in dB). Typical equations include only one element in dBm on each side (modified by any number of ratios in dB), only ratios in dB, or differences of two dBm values (which become dB ratios). One of the simplest dB form equations is illustrated by the amplifier in Figure 2-1, which multiplies input signals by a gain factor.

The linear form of the amplifier equation is

\[ P_O = P_I \times G \]

where \( P_O \) is the output power, \( P_I \) is the input power, and \( G \) is the gain of the amplifier.

Both power numbers are in linear units (e.g., milliwatts), and \( G \) is the gain factor in linear form (e.g., 100). If the input power is 1 milliwatt, an amplifier gain of 100 will cause a 100 milliwatt output signal.

By converting the input power to dBm and the gain to dB, the equation becomes

\[ P_O = P_I + G \]

The output power is now expressed in dBm. Using the same numbers, 1 milliwatt becomes 0 dBm, the gain becomes 20 dB, and the output power is +20 dBm. (This can be converted back to 100 milliwatts in linear units if required).

![Figure 2-1 Amplifier](image-url)
This is a very simple case, in which the marginally simpler calculation does not seem worth the trouble to convert to and from dB forms. But now consider a typical communication theory equation. As will be shown in Chapter 5, a transmitted signal is reduced by a spreading loss that is proportional to the square of its frequency, $F$, and the square of the distance, $d$, it travels from the transmitting antenna. Thus, the spreading loss is the product of $F^2$, $d^2$, and a constant (which includes several terms from the derivation). The formula is then

$$L = K \times F^2 \times d^2$$

In dB form, $F$ (dB) becomes $10 \log(F)$. $F^2$ is $2[10 \log(F)]$ or $20 \log(F)$, and $d^2$ is transformed to $20 \log(d)$ the same way. The constant is also converted to dB form, but first it is modified with conversion factors to allow us to input values in the most convenient units and generate an answer in the most convenient units. In this case, $K$ is multiplied by the necessary conversion factors to allow frequency to be input in MHz and distance to be input in kilometers. When the log of this whole mess is multiplied by 10, it becomes 32.44, which is commonly rounded to 32. The spreading loss in dB can then be found directly from

$$L_S = 32 + 20 \log(F) + 20 \log(d)$$

where $L_S$ is spreading loss (in dB), $F$ is frequency (in MHz), and $d$ is distance (in km), which most people find much easier to use in practical applications. In later chapters, dB equations for all of the communication link relationships will be presented.

It is important to understand the role of the constant in this type of equation. Since it contains unit conversion factors, this equation works only if you input values in the proper units. In this book, the units for each term are given right below the dB equation—every time. You will memorize some of these equations and use them often; be sure you also remember the applicable units.

### 2.4 Quick Conversions to dB Values

Table 2-1 gives some common dB values with their equivalent linear ratios. For example, multiplying a linear number by a factor of 1.25 is the same as adding 1 dB to the same number in dB form. (1 milliwatt × 1.25 is the same as 0 dBm + 1 dB so 1.25 milliwatts = 1 dBm).

<table>
<thead>
<tr>
<th>Ratio</th>
<th>dB Value</th>
<th>Ratio</th>
<th>dB Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>1/10</td>
<td>−10</td>
<td>1.25</td>
<td>+1</td>
</tr>
<tr>
<td>1/4</td>
<td>−6</td>
<td>2</td>
<td>+3</td>
</tr>
<tr>
<td>1/2</td>
<td>−3</td>
<td>4</td>
<td>+6</td>
</tr>
<tr>
<td>1</td>
<td>0</td>
<td>10</td>
<td>+10</td>
</tr>
</tbody>
</table>
This table is extremely useful because it will allow you to make quick determinations of approximate dB values without touching a calculator. Here’s how it works:

- First get from one (1) to the proper order of magnitude. This is easy because each time you multiply the linear value by 10 you just add 10 dB to its dB value. Likewise, dividing by 10 subtracts 10 dB from the dB value.
- Then use the ratios from Table 2-1 to get close to the desired number.

For example, 400 is $10 \times 10 \times 4$. In dB form, these manipulations are $10 \, \text{dB} + 10 \, \text{dB} + 6 \, \text{dB}$. A more common way to look at the manipulation is

$$400 = 20 \, \text{dB} \, \text{(which gets you to 100)}$$
$$+ 6 \, \text{dB} \, \text{(to multiply by 4)} = 26 \, \text{dB}$$

$$500 \approx 30 \, \text{dB} \, \text{(= 1000)}$$
$$– 3 \, \text{dB} \, \text{(to divide by 2)} = 27 \, \text{dB}$$

Be careful not to be confused by 0 dB. A high-ranking government official once embarrassed himself in a large meeting by announcing: “The signal is completely gone when the signal-to-noise ratio gets down to 0 dB.” In fact, a 0 dB ratio between two numbers just means that they are equal to each other (i.e., have a ratio of 1).

Table 2-2 shows the power in dBm for various linear power values. This is a very useful table, and we will use these values many times in examples in later chapters.

Other values often expressed in dB form are shown in Table 2-3.

There is one more chance to be seriously confused about dBs. Voltage ratios are often expressed in dB, but it is common to convert the linear voltage ratio to dB using

$$\text{Voltage Ratio (in dB)} = 20 \log(\text{Linear Voltage Ratio})$$

The basis for this practice is that the ratio of two power levels is equivalent to the ratio of the squares of two voltage levels (because $P = V^2/R$). A voltage ratio will be expressed in dB only once in this book (on page 107).

<table>
<thead>
<tr>
<th>dBm</th>
<th>Signal Strength</th>
<th>dBm</th>
<th>Signal Strength</th>
</tr>
</thead>
<tbody>
<tr>
<td>+90</td>
<td>1 megawatt</td>
<td>+20</td>
<td>100 milliwatts</td>
</tr>
<tr>
<td>+80</td>
<td>100 kilowatts</td>
<td>+10</td>
<td>10 milliwatts</td>
</tr>
<tr>
<td>+70</td>
<td>10 kilowatts</td>
<td>0</td>
<td>1 milliwatt</td>
</tr>
<tr>
<td>+60</td>
<td>1 kilowatt</td>
<td>–10</td>
<td>100 microwatts</td>
</tr>
<tr>
<td>+50</td>
<td>100 watts</td>
<td>–20</td>
<td>10 microwatts</td>
</tr>
<tr>
<td>+40</td>
<td>10 watts</td>
<td>–30</td>
<td>1 microwatt</td>
</tr>
<tr>
<td>+30</td>
<td>1 watt</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Table 2-3  Common dB Definitions

<table>
<thead>
<tr>
<th></th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>dBm</td>
<td>= dB value of ( \frac{\text{Power}}{1 \text{ milliwatt}} )</td>
</tr>
<tr>
<td>dBW</td>
<td>= dB value of ( \frac{\text{Power}}{1 \text{ watt}} )</td>
</tr>
<tr>
<td>dBsm</td>
<td>= dB value of ( \frac{\text{Area}}{1 \text{ square meter}} )</td>
</tr>
<tr>
<td>dBi</td>
<td>= dB value of antenna gain relative to that of an isotropic antenna**</td>
</tr>
</tbody>
</table>

*Commonly used for Antenna Area and Radar Cross Section.
**See Section 4.4.1.

2.5 dB Bookkeeping Diagrams

It is easy to become confused when manipulating gains, losses, and signal-strength levels in the various communications equations. Since the dB forms of equations typically involve addition and subtraction, they can be illustrated using simple diagrams that many people (enthusiastically including the author) find helpful in avoiding confusion. Through the rest of the book, small dB bookkeeping diagrams like this one will be inserted in the paragraphs in which the equations they represent are discussed. Wherever practical, the diagrams will be placed right with the equations they illustrate, as shown in Figure 2-2.

![Figure 2-2 dB equation with “book keeping” diagram](image)

These diagrams will not be dignified with figure numbers and usually will not be discussed. Their purpose is to help you keep what is happening in perspective.

**Figure 2-2 dB equation with “book keeping” diagram**

\[
\text{ERP} = P_T + G_T
\]

where:
\[
P_T = \text{Transmitter power in dBm}
\]
\[
G_T = \text{Transmit antenna gain in dB}
\]
In every such diagram:

- Equation terms that are signal strength values are represented by horizontal lines.
- Equation terms that cause changes in signal strength are vertical arrows (pointing in the direction that the signal strength changed by that term of the equation).
- The action in the diagram moves from left to right as you move through the right side of the equation.
- If the answer term of the equation (on the left side) is a ratio of signal strengths, it is shown as a vertical double-headed arrow.
- If the answer term is a signal strength, it is shown as the final horizontal line in the diagram.
- All signal-strength levels are in dBm, and all changes in signal strength levels are in dB.
- No attempt is made to draw the signal strength values to scale; only the relative positions are significant.

Some people really like these diagrams, and some people don’t. If you are in the latter group, please feel free to ignore them in good health. You purchased this book, and the customer is always right.
Chapter 3

Signal-to-Noise Ratio

In considering communication link performance, signal-to-noise ratio (SNR) is an extremely important concept. It is the common way to quantify the quality of the signal at any point in the communication process. Ultimately, it is the SNR that determines whether or not adequate communication takes place, but as will be seen in later chapters its requirement at each point in the link depends on many factors.

3.1 Noise

Whole books are written on the nature of noise and its mathematical characterization, but it is sufficient for our purposes to define noise as random signals that are present along with desired signals in any communication channel. In many specifications, any nonrandom but undesired signals present in a circuit are also considered to be noise—particularly if they cannot be predicted or if several intermodulate in interesting ways. In this book, we won’t call them noise unless they are noncoherent and must be considered present over a bandwidth greater than that in which we measure them.

We will consider two types of noise: thermal and man-made. Thermal noise comes from molecular activity in the atmosphere and in the materials from which the receiver is manufactured. The power of the thermal noise in an ideal circuit is calculated by the expression $kT/2$. As discussed in detail in Chapter 6, $kT/2$ is the product of Boltzmann’s constant, the temperature (in Kelvin), and the bandwidth in which it is measured. Chapter 6 also discusses thermal noise in nonideal circuits.

There are always interfering signals from inside and outside the receiver. These include ripple from power supply voltages and conducted and radiated electromagnetic interference (EMI). Good design practice normally calls for all of these undesired signals to be lower than the thermal noise. However, if that is impossible or impractical, they will diminish the quality of the signal being communicated. While they are not literally noise, they are often difficult to distinguish from noise and are therefore often included in the noise part of the SNR as discussed already.

External noise and interference signals are received by the receiving antenna along with desired signals. Interfering signals (deliberate or accidental) are discussed in Section 5.4. They are not considered in the noise part of the SNR and must usually be dealt with individually.
External noise is the sum of all of the radio frequency (RF) emissions, both natural and man-made, that reach the receiving antenna (except those we consider individually as interfering signals). If the receiver is very near a large noise generator (e.g., the engine in a motor vehicle) this particular source can be (and usually must be) reduced by filtering at its source or avoided through some operational approach (like turning the engine off when we want to operate the receiver). A level of general background noise is also always present. Section 3.4 includes a chart quantifying this noise as a function of frequency and type of location (urban, suburban, or rural).

### 3.2 Signal-to-Noise Ratio

The SNR is defined as the ratio of the signal power to the noise power (from all sources). The absolute levels of signal and noise are normally stated in dBm, so the ratio of the two levels (the signal to noise ratio) is in dB. For example, if the signal power ($S$) were –104 dBm and the noise power ($N$) were –114 dBm, the SNR would be 10 dB. The SNR in dB is the difference of the signal and the noise powers (rather than the quotient) because both are in dBm (i.e., logarithmic numbers).

![Figure 3-1 Signal with noise in time domain](image)

In the real world, there is no such thing as a pure signal—you can only measure a signal along with some level of noise. When you think you are seeing a pure signal, it is just that the SNR is so large that you cannot detect the noise.

Figure 3-1 shows a signal with noise: in the time domain, as it would appear on an oscilloscope. The noise appears fuzzy on the oscilloscope because it is jittering...
in both time and amplitude from sweep to sweep. The image of noise on an oscilloscope is often called the grass.

This figure shows a fairly large SNR, which means that the signal is easily processed. If an audio signal had the SNR depicted, the noise would be perceptible to a listener but not objectionable—good enough to understand speech but less than high fidelity.

Figure 3-2 shows the same signal in the frequency domain (as it would appear on a spectrum analyzer). Again, the noise part of the display will appear fuzzy because it jitters in both frequency and amplitude.

Figure 3-3 shows a much lower SNR (in the time domain), and Figure 3-4 shows the same signal with its associated noise in the frequency domain.
In a communication system, the SNR of the received signal can be no better than that in the transmitted signal, but transmitters are designed to output very high SNRs, so we will be mainly concerned about noise added to the signal by the receiver itself and transmitted into the receiving antenna from other sources.

3.3 Other Related Ratios

Some types of modulation provide improvement in SNR for output signals—relative to the SNR of the RF signals that are received. Two important examples are frequency modulated signals and digital signals. In both cases, two signal-to-noise values must be considered: the output SNR and the RF SNR, which can be significantly different.

3.3.1 RF SNR

For frequency modulated or digitized signals, the SNR is normally taken to mean the output audio or video SNR. In this book, the SNR of the actual RF signal received is called the RF SNR. Some textbooks call it the carrier-to-noise ratio (CNR) (Figure 3-5). As will be shown in Chapter 6, the output SNR of frequency modulated signals can be greatly improved from the RF SNR as a function of the frequency modulation index as long as the RF SNR is above a threshold level. The output SNR of digitally modulated signals is primarily the signal-to-quantization ratio (SQR) and is only secondarily related to the RF SNR.

3.3.2 Signal-to-Quantization Ratio

The SQR is the ratio of the signal amplitude to the quantization increment as shown in Figure 3-6. When an analog signal (e.g., a voice channel) is digitized, an analog-to-digital converter (ADC) compares it to a series of thresholds and generates a set of digital signals (ones and zeros) to describe the time history of the waveform digitized. The signal is transmitted as a digital bit stream modulated onto an RF signal. In the receiver, the RF signal is demodulated to reproduce the digital bit stream. When the analog signal is reconstructed (from the digital bit stream) by a digital-to-analog converter (DAC), it is not a smooth curve but jumps between defined levels as shown in Figure 3-6. The number of these levels is a function of the number of bits used in the digitization of each signal level. The nonsmoothness

---

*Figure 3-5 RF SNR or CNR vs. SNR*
of the reconstructed signal is usually called quantization noise. The output signal quality is then properly specified as the signal-to-quantization noise ratio; however, it is often called just the signal-to-noise ratio.

It is important to understand that this quantization noise is completely independent of the SNR of the received RF signal. As will be shown in Chapter 6, the RF SNR determines the bit error rate in the received digital signal, but the SQR is set in the original digitizing process. One of the great charms of digital communication systems is that the SQR remains constant even if the signal is subjected to many processing steps, whereas nondigitized signals tend to be degraded by every processing step. Section 6.3.3 includes a formula to determine signal to quantization noise ratio as a function of the number of bits per sample.

Naturally, the signal cannot be properly reproduced by the receiver unless it detects the proper bits, so the bit error rate is also important and depends on the RF SNR. Section 6.3.3 presents curves of the bit error rates as a function of $E_b/N_0$, which is a variable based on both the RF SNR and the ratio of the receiver effective bandwidth to the signal bit rate.

### 3.4 Background Noise

Figure 3-7 is a characterization of the total, nonthermal background noise that will be seen by an omnidirectional antenna in various types of locations. This is the electronic noise from street cars, automobile and truck engines, electric motors,
high-order intermodulations of hundreds of unrelated radio transmissions, and noise from outer space.

The chart shows the expected background noise in terms of dB above ideal thermal noise in a receiver of any specified bandwidth (called dB above kTB in Figure 3-7). Note that the noise levels in this chart are independent of the receiver bandwidth because kTB has a bandwidth term.

In Chapter 6 (Section 6.1) formulas and a graph are provided to determine the actual noise power of kTB in terms of temperature and bandwidth. After calculating kTB, the level of background noise power reaching an antenna can be determined from the appropriate lines on the graph.

The three lines on the chart show the expected noise power level for urban locations, suburban locations, and very quiet rural locations in which the noise is primarily cosmic.

It must be emphasized that these curves are average values based on two different sources that combined data from many surveys and with much smoothing of data. (The data vary between sources.) All of the source data are collected using omnidirectional antennas. While impossible to predict with precision, the expected background noise can be an important system design parameter, particularly at low
frequencies, where it may become dominant over the internal system noise of very sensitive receivers.

For an example calculation of the background noise level, consider a receiver with a 1 MHz bandwidth. Its kTB level (see Chapter 6) would be –114 dBm. If operated in a noisy urban area at 100 MHz using an omnidirectional antenna, this receiver would expect to receive a background noise level of –76 dBm (–114 dBm + 38 dB).
Chapter 4
The Communication Link

The communication link is everything required to get information signals from one point to another without wires. In its simplest form, as shown in Figure 4-1, the link includes a transmitter, a receiver, transmitting and receiving antennas, and everything that happens to the signals between the two.

This chapter provides functional descriptions of each of these link elements and also of the information signals that links are designed to carry.

4.1 Information Signals

The information signal carries the information to be communicated over the link. Information signals come in a bewildering variety of forms, but the most common are audio signals, digital data, and video signals.

Each of these types of information signals can be characterized by its information bandwidth and the output signal-to-noise ratio (SNR) it requires to assure its adequate communication to the information user at the other end of the link. Table 4-1 shows each of these basic types along with some common subtypes and the demands each places on the communication link.

4.1.1 Audio Signals

Audio signals typically include human voice, music, or other audible sounds. They are limited in bandwidth to match the frequency response of the human ear.

Voice signals for communication must have adequate bandwidth for a human to receive the information passed. The most common example is a telephone circuit, which typically accepts an input band from 300 to 3,400 Hz. This frequency range allows us to easily understand what is being said and to recognize the voice of the speaker, but it clearly limits the quality of the output and is not nearly adequate to pass music. Military communication networks can, if necessary, use even narrower bandwidth because of the disciplined vocabulary used by military communicators.

Music requires adequate bandwidth to reproduce the tonal qualities of singing voices and musical instruments. High fidelity music transmission requires a frequency range covering from less than 10 Hz to above 15 kHz for concert hall realism.

A special case audio signal (the first used in radio communication) is single-tone Morse code. Morse code signals are typically generated by turning the radio...
frequency (RF) carrier on and off. This is called on-off-keyed (OOK) modulation, which is very narrow in bandwidth. The actual beeps heard by the receiving operator are generated in the receiver when the signal is present.

4.1.2 Digital Data

Digital data are a series of ones and zeros that form binary codes to represent numbers, letters, and graphical data. Computers talk to each other digitally, but many other devices also input or output controls or data in digital form. Each one or zero is a bit. Although digital data can be formatted in many ways, they are normally collected into 8-bit bytes. These are typically gathered into frames and subframes that include synchronization provisions to allow the user of the data to identify individual bits and bytes in long data streams (Figure 4-2). The synchronization signals are most often transmitted as extra bits. These synchronization

![Figure 4-1 Elements of the communication link](image_url)

### Table 4-1 Information Signals

<table>
<thead>
<tr>
<th>Type of Signal</th>
<th>Typical Info Bandwidth</th>
<th>Typical SNR Required</th>
</tr>
</thead>
<tbody>
<tr>
<td>Audio (Voice)</td>
<td>4 kHz</td>
<td>10 to 20 dB</td>
</tr>
<tr>
<td>Audio (Music)</td>
<td>15 kHz</td>
<td>20 to 40 dB</td>
</tr>
<tr>
<td>Digital Data</td>
<td>&lt;1 Hz to 1 GHz</td>
<td>10 to 20 dB</td>
</tr>
<tr>
<td>Video (Analog Television)</td>
<td>4 MHz</td>
<td>40 dB or more</td>
</tr>
<tr>
<td>Video (Digital Television)</td>
<td>6 MHz (includes other info)</td>
<td>10 to 20 dB</td>
</tr>
<tr>
<td>Video (Multichannel)</td>
<td>20 kHz to 4 GHz</td>
<td>10 to 20 dB</td>
</tr>
<tr>
<td>Video (Pulse)</td>
<td>.1 to 10 MHz</td>
<td>10 to 20 dB</td>
</tr>
</tbody>
</table>

![Figure 4-2 Digital data stream](image_url)

Although digital data can be formatted in many ways, they are normally collected into 8-bit bytes. These are typically gathered into frames and subframes that include synchronization provisions to allow the user of the data to identify individual bits and bytes in long data streams (Figure 4-2). The synchronization signals are most often transmitted as extra bits. These synchronization
bits—as well as others that are added to designate an intended user or to allow error detection and correction—are often referred to as overhead since the transmission medium must have enough throughput capacity to pass them, but they carry no direct signal information.

For transmission, digital data are typically formatted in serial form, that is, one bit at a time (rather than in parallel form with each bit in a byte having its own wire, as is often the case inside of hardware). However, in more complex modulation schemes such as quadrature phase-shift keying (QPSK), more than one simultaneous bit can be present in a single signal.

Digital signals have widely varying bandwidth, from much less than 1 Hz to gigabit rates (more than $10^9$ bits/sec). The signal bandwidth depends on the bit rate, which is in turn dictated by the amount and fidelity of information that must be transmitted per unit time.

### 4.1.3 Video Signals

Video signals convey information that must be changed to another form before it can be perceived by people or computers. The most common video signals carry broadcast television picture information. These signals have 4 MHz bandwidth and contain all of the information required to allow synchronization of the TV frame and to set the brightness and color of each dot on the screen. Figure 4-3 shows the structure of the analog television signal as a function of frequency, and Figure 4-4 shows it as a function of time. Digital television uses the same 6 MHz total channel bandwidth but adds other information along with the compressed video signal.

Another type of video signal contains many audio channels that are multiplexed into one signal so they can be transmitted together (Figure 4-5). These multichannel signals are used in microwave telephone links and a few other applications. The video bandwidth is typically 4 kHz per voice channel slot. The individual voice grade channels are input to a modulator, which creates the video signal to drive the transmitter, and are demodulated and distributed after being output from a receiver.

![Figure 4-3 Television video signal in frequency domain](image-url)
A third type of video signal is pulsed. As shown in Figure 4-6, pulses are short duration signals that are used to abruptly turn the transmitter on and off, generating short bursts of RF energy. The most common use of pulsed signals is in radars. Typical information bandwidths for pulse signals are from 100 kHz to 10 MHz.

The bandwidth of a pulse signal is determined by the width of the pulse: the shorter the pulse, the wider the bandwidth. Adequate bandwidth is often taken to be
one over the pulse width; for example, a 1 µsec long pulse would require 1 MHz of bandwidth. However, the full spectrum occupancy of a nice square pulse includes many harmonics. The actual transmission bandwidth required will depend on the performance specifications of the radar generating the signal.

The receiver can narrow the bandwidth to improve sensitivity, but the shape of the received pulse depends on the narrowest filter it passes through. Figure 4-7 shows the shape of a pulse after various levels of filtering. Any of these shapes may be optimum depending on the situation and the postdetection processing applied to the signal.

4.2 Transmitters

Transmitters accept information inputs from various sources (e.g., people, computers, sensors) and convert them into transmittable forms. The transmitter output
is characterized by its transmission bandwidth (in some units of frequency) and its power level (which is often stated in watts or kilowatts but can be converted into dBm for convenience in link performance predictions).

Figure 4-8 shows a block diagram of a generalized transmitter. The actual circuitry required to implement each of these blocks varies widely and is covered in detail in many communication theory texts. Our purpose here is just to deal with them functionally and to describe what each block does rather than how it does it.

The oscillator generates a carrier signal onto which the input information is modulated. The oscillator sets the RF of the transmitted signal (often called the RF frequency to distinguish it from modulating frequencies).

The use of radio frequencies is controlled by a set of national and international agreements to minimize interference among the many people trying to communicate. This is necessary because two or more signals at the same frequency in the same general area will interfere with each other.

Transmission frequencies are broken into bands that have a variety of names, as shown in Figure 4-9. These band names have been developed as a convenient way to specify and discuss hardware (e.g., an S-band amplifier, a J-band radar). When in doubt, it is best to specify an RF frequency or frequency range directly in units of frequency (Hz, kHz, MHz, or GHz). Table 4-2 describes the typical communications uses and characteristics of the various frequency ranges.

4.2.1 Modulator

The modulator modulates the input information signal onto the RF carrier for transmission. The main characteristics of the modulation that affect the communication link performance are the amount of frequency spectrum that the modulated signal occupies and any signal-to-noise improvement factor.

The following types of modulation are the most common. Each is described in terms of its typical applications and its impact on link performance. There are many other modulation schemes, but most can be quite accurately characterized by their similarity to those described here.
Amplitude Modulation

Amplitude modulation (AM), as shown in the time domain in Figure 4-10, carries the information signal as a series of variations in the amplitude (hence the transmitted power level) of the carrier. Figure 4-11 shows an AM signal in the frequency domain. The information signal is carried in two sidebands, which causes the spectrum occupancy to be approximately twice the information bandwidth. A detected AM signal typically provides an SNR equal to the RF SNR.
<table>
<thead>
<tr>
<th>Frequency Range</th>
<th>Abbreviation</th>
<th>Type of Communication and Characteristics</th>
</tr>
</thead>
<tbody>
<tr>
<td>Very Low, Low, and Medium VHF</td>
<td>VHF</td>
<td>Very long range communication (e.g., ships at sea)</td>
</tr>
<tr>
<td>Frequency 3 kHz to 3 MHz</td>
<td>LF</td>
<td>Commercial AM radio</td>
</tr>
<tr>
<td>High Frequency 3 to 30 MHz</td>
<td>MF</td>
<td>Ground waves circle Earth</td>
</tr>
<tr>
<td>Very High Frequency 30 to 300 MHz</td>
<td>VHF</td>
<td>Over-the-horizon communication</td>
</tr>
<tr>
<td>Ultra High Frequency 300 MHz to 1 GHz</td>
<td>UHF</td>
<td>Mobile communication, TV, and commercial FM radio</td>
</tr>
<tr>
<td>Microwave 1 to 30 GHz</td>
<td>mw</td>
<td>Line of sight required, but knife edge diffraction can extend range</td>
</tr>
<tr>
<td>Millimeter Wave 30 to 100 GHz</td>
<td>MMW</td>
<td>TV and telephone links, satellite communication, radar</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Line of sight required</td>
</tr>
<tr>
<td></td>
<td></td>
<td>High-resolution radars</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Requires line of sight</td>
</tr>
<tr>
<td></td>
<td></td>
<td>High absorption in rain and fog</td>
</tr>
</tbody>
</table>

Figure 4-10  Amplitude modulation

Figure 4-11  AM signal in frequency domain
Frequency Modulation

Frequency modulation (FM), as shown in the time domain in Figure 4-12, carries the information signal as a series of variations in the frequency of the carrier. Figure 4-13 shows an FM signal in the frequency domain. Like AM, FM signals have two information-carrying sidebands, but they can occupy much more frequency spectrum because the amount of frequency variation can be several times the information bandwidth. The ratio of the transmitted frequency variation to the modulating frequency is called the modulation index ($\beta$). The ratio of the peak frequency deviation to the information bandwidth is also $\beta$.

The total frequency band required to transmit an FM signal is a nonlinear function of the modulation index but is typically set at

$$BW_{FM} = 2 \times \text{Information Signal Bandwidth} \times \text{Modulation Index}$$

since this transmission bandwidth provides adequate quality for almost all types of information signals.

The greater the modulation index, the greater the noise and interference immunity of the transmitted signal. The output SNR of a demodulated FM signal is greater than the RF SNR by a factor related to the modulation index, as described in Chapter 6.

![Figure 4-12 Frequency modulation](image)

![Figure 4-13 FM signal in frequency domain](image)
Single Sideband

Single sideband signals are in effect AM signals with the carrier and one sideband filtered off. Although they are more complex to generate and demodulate than AM signals, they occupy about half of the spectrum width. The spectrum occupancy of a single sideband signal equals the information bandwidth, and the RF SNR is equal to the demodulated SNR.

Modulations for Digital Signals

Before they can be transmitted, digital signals must be amplitude modulated (Figure 4-14), phase modulated (Figure 4-15), or frequency modulated onto an RF carrier.

The simplest frequency modulation scheme is frequency shift keying (FSK), in which one frequency represents a digital zero and another represents a digital one (Figure 4-16). The frequency spectrum has more than just those two spectral lines, however, because of the sudden frequency change that occurs when the digital signal switches from a zero to a one.

The RF spectral occupancy of a digital signal depends on two factors: the digitization and the modulation. Figure 4-17 shows the spectrum of an RF signal carrying digital information, and Figure 4-18 describes the important parameters of that spectrum. The null-to-null bandwidth of this spectrum in Hz is typically twice the bit rate in bits per second. The 3 dB bandwidth is 0.88x the bit rate for most modulations. Note that this bit rate includes all of the overhead bits as well as the data being transmitted.

---

**Figure 4-14**  Amplitude modulated digital signal

**Figure 4-15**  Phase modulated digital signal
Figure 4-16  FSK modulated digital signal

Figure 4-17  Spectrum of a digital transmission

Figure 4-18  Digital spectrum parameters
Minimum shift keying (MSK) is a particularly efficient modulation technique used in many systems. Its null-to-null bandwidth is only 1.5x the bit rate, and its 3 dB bandwidth is 0.66x the bit rate. Several complex digital modulations carry multiple information bits (e.g., 4 bits) in each transmitted baud. These modulations are sent with each baud at a specific amplitude and phase. When a baud is received at one of the available amplitude/phase values, the receiver outputs the appropriate bits.

**Digitization**

Analog signals (e.g., voice or music) are converted to digital form by digitization in an analog-to-digital converter (ADC). The required number of bits per unit time (the bit rate) is determined by the following factors:

- The sample rate, the rate at which the input waveform must be sampled to allow accurate reproduction after demodulation, is usually twice the highest frequency contained in the information signal. This is called the Nyquist rate.
- The digitization level, the number of bits per sample, must be great enough to provide adequate signal to quantizing noise ratio.
- The overhead, the number of bits that are added to the bit stream to provide synchronization, addresses, parity checks, and error correcting codes, typically varies from 10% to more than 100% of the bits dedicated to the information being transmitted. The highest overhead rates are associated with systems using extensive error detection and correction schemes.

The total bit rate (in bits per second) is

\[
Bit Rate = 2 \times Information\ Bandwidth \times Sample\ Rate \times (1 + Overhead\ Factor)
\]

In many applications, the transmitted information is originally digital, for example when one computer communicates with another. In this case, the total bit rate is

\[
Bit Rate = Information\ Bit\ Rate \times (1 + Overhead\ Factor)
\]

Communication system specifications are often written in terms of byte rate (usually one byte = 8 bits, but not always) or message rate (a standard message being some fixed number of bytes). To determine the transmission bandwidth required or to predict the link performance, it is necessary to convert either of these to the equivalent bit rate.

**Modulation for Transmission**

Various techniques are used to modulate the digital bit stream onto an RF carrier for transmission. Each has advantages and disadvantages that depend on the specific application. When viewed from the perspective of communication link throughput performance, each can be characterized in terms of the amount of RF bandwidth required to transmit a specified bit rate.

Common practice is to filter the RF modulated signal so the 3 dB bandwidth is transmitted. As mentioned earlier, this is approximately 0.88x the bit rate.
For most types of modulations commonly used, this yields a ratio of RF bandwidth (in Hz) to bit rate in bits per second (bps), which varies over the range one to two.

\[ RF \text{ Bandwidth} \text{ (typically)} = 1 \text{ to } 2 \text{ Hz/bps} \]

Table 4-3 shows several popular modulation schemes, along with the RF frequency occupancy factor each requires. This is important information because digital links are often specified in terms of their digital throughput characteristics. From Table 4-3 you can see that the choice of a frequency-efficient modulation approach dictates an RF bandwidth (in Hz) numerically equal to the bit rate (in bps). There are, however, some highly efficient codes that carry more bits per second per Hz of bandwidth, but they require greater received SNRs.

This will allow you to use the formulas in Chapters 5 and 6 of this book to predict digital link performance.

### 4.2.2 Power Amplifier

Modulation normally is done at low power levels. Then the power amplifier increases the signal power to the level required to achieve the required transmission range and outputs a signal to the transmitting antenna. The power amplifier output power can range from few milliwatts to many kilowatts. This power is stated in dBm for link analysis purposes. Since the total radiated power is the transmitter power plus (in dB) the antenna gain, there is a trade-off between transmit antenna gain and transmitter power.

The maximum transmitter output power available depends on the type of power amplifier used, the signal duty cycle, and the frequency range.

Solid-state amplifiers are lower in power output and are restricted to lower frequency ranges than various types of tube amplifiers, but they have been significantly more reliable in operation, particularly in rugged environments. Both of these factors change constantly, though. A great deal of work is being done to improve tube reliability, and both the maximum power and frequency range of solid-state amplifiers are constantly being increased.

The duty cycle of the signal makes a significant difference in the maximum output power available. The peak power of a continuous wave (CW) signal is equal to its average power. However, the average power of a pulsed signal can be two or three orders of magnitude less than its peak power.

---

**Table 4-3 Bandwidth Factors for Various Modulations Used for Digital Data**

<table>
<thead>
<tr>
<th>Modulation</th>
<th>Bandwidth Factor (Hz/bps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Frequency Shift Keying (FSK)</td>
<td>2</td>
</tr>
<tr>
<td>Binary Phase-Shift Keying (BPSK)</td>
<td>0.88</td>
</tr>
<tr>
<td>In-Phase and Quadrature (I/Q)</td>
<td>0.88/(bits/aud)</td>
</tr>
<tr>
<td>Minimum Shift Keying (MSK)</td>
<td>0.66</td>
</tr>
</tbody>
</table>

(or 0.66x for MSK).
Although the state of the art is continually changing, CW amplifiers with greater than 1 kW of output have been hard to find for a long time, whereas pulse amplifiers capable of hundreds of kilowatts have been common for many years.

### 4.3 Receivers

The receiver performs the following link functions:

- Eliminates all but the desired signal (or signals)
- Demodulates the received signal to reconstruct the original information signal input to the transmitter
- Band-limits the received signal for sensitivity

Many different types of receivers have been developed to perform these functions, and the techniques vary depending on the frequency range, the type of signals to be received, and other operational considerations. Table 4-4 shows a few of the many types of receivers in common use, along with the characteristics of each. These few types have been chosen for discussion to facilitate later discussions of the impact of the receiver on link performance. The relative merits of different types of receivers are usually described in terms of their selectivity and sensitivity.

**Table 4-4 Various Types of Receivers**

<table>
<thead>
<tr>
<th>Receiver Type</th>
<th>Characteristics</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fixed Tuned</td>
<td>Good selectivity and sensitivity</td>
</tr>
<tr>
<td></td>
<td>Dedicated to one signal</td>
</tr>
<tr>
<td>Channelized</td>
<td>Combines selectivity and sensitivity with wideband coverage</td>
</tr>
<tr>
<td>Superheterodyne</td>
<td>Most common type of receiver</td>
</tr>
<tr>
<td></td>
<td>Good selectivity and sensitivity</td>
</tr>
<tr>
<td>Crystal Video</td>
<td>Wideband instantaneous coverage</td>
</tr>
<tr>
<td></td>
<td>Low sensitivity and no selectivity</td>
</tr>
<tr>
<td></td>
<td>Mainly for pulsed signals</td>
</tr>
<tr>
<td>Instantaneous Frequency</td>
<td>Sensitivity like crystal video</td>
</tr>
<tr>
<td>Measurement (IFM)</td>
<td>Covers up to 1 octave</td>
</tr>
<tr>
<td></td>
<td>Measures frequency only</td>
</tr>
<tr>
<td></td>
<td>Can have only 1 signal in band</td>
</tr>
<tr>
<td>Compressive Receiver</td>
<td>Sensitivity like superhet</td>
</tr>
<tr>
<td></td>
<td>Covers up to one octave</td>
</tr>
<tr>
<td></td>
<td>Measures frequency only</td>
</tr>
<tr>
<td></td>
<td>Can handle multiple simultaneous signals</td>
</tr>
<tr>
<td>Digital</td>
<td>Highly flexible</td>
</tr>
<tr>
<td></td>
<td>Can deal with signals with unknown parameters</td>
</tr>
<tr>
<td></td>
<td>Significant processing capability</td>
</tr>
</tbody>
</table>
Selectivity is the ability to ignore undesired signals or interference and is typically related to the instantaneous bandwidth covered. Sensitivity defines the weakest signal that can be received. The characteristics described in this table are generalities that are useful in selecting a type of receiver to perform a particular task. However, it must be recognized that the selectivity and sensitivity of an individual receiver of any type can vary over a wide range.

4.3.1 Fixed Tuned Receiver

The fixed tuned receiver (Figure 4-19) typically monitors a single transmitter or an important broadcast frequency. The tuner normally includes a fixed tuned pre-selection filter to limit the input to the single frequency of interest. The demodulator is appropriate for the signals to be received, and the recovered information signal can either be output to a listener or further processed.

Fixed tuned receivers typically have high sensitivity and good selectivity. They are also usually small, light, reliable, and relatively inexpensive—all because they have relatively few components. However, they are necessarily lacking in flexibility because they are not tunable.

Examples of practical fixed tuned receivers are single-frequency communication receivers, emergency broadcast channel monitors, weather report receivers, and paging receivers. A very interesting application of a fixed tuned receiver is for receipt of global positioning system (GPS) signals, which come from many different satellites, four or more of which are usually within view of any receiver. However, each signal contains a different code that uniquely spreads its spectrum. By applying the correct code, the GPS receiver reconstructs an individual signal and effectively tunes out all other signals. The scheme is called code division multiple access (CDMA) and was selected for several reasons, including the practicality of using simple (and economical) fixed tuned receivers.

4.3.2 Channelized Receiver

To monitor large frequency ranges, several fixed tuned receivers can be organized into a channelized receiver (Figure 4-20) to demodulate the whole frequency range with high sensitivity. The filter bank normally divides the frequency range being covered into contiguous channels; that is, the filters for adjacent channels have the upper 3 dB frequency of the lower channel equal to the lower 3 dB frequency of the upper channel.
The big advantage of a channelized receiver is that it can simultaneously look at and demodulate signals over its full frequency range with high sensitivity and selectivity. The output selection circuit can then output any appropriate number of demodulated signals for recording or further processing. The disadvantage of the channelized receiver is its high complexity and cost relative to other types of receivers. State of the art packaging techniques are making channelized receivers more and more practical. However, channelized receivers remain significantly more complex and expensive than most other types of receivers, so their use is limited to applications that require high probability of receiving large numbers of signals, some of which are simultaneous. The best examples are in the electronic warfare field, where receiving many signals in an extremely short time frame is literally a matter of life and death.

As a practical matter, channelized receivers are usually used in combination with switchable band selectors that operate on the superheterodyne principle so that a single channelized receiver with a reasonable number of channels can be quickly assigned where it is needed over a large frequency band.

4.3.3 Superheterodyne Receiver

The superheterodyne (commonly called superhet) receiver is the most widely used. Its great charm is that it can provide virtually any type of performance that is required (within the limits of the laws of physics, of course). It allows infinite trade-offs of sensitivity and selectivity against band coverage and tuning speed. It can cover relatively wide frequency ranges with relatively low complexity. For these reasons, commercial AM, FM, and short-wave broadcast receivers are almost universally superhets.
Figure 4-21 shows the basic block diagram of a superhet single-conversion receiver. Actual superhet receivers often have significantly greater complexity, including two or three frequency conversions to cover all of the required frequency range with acceptable received signal quality. They may also have several selectable intermediate frequency (IF) amplifiers (with different bandwidths) or multiple demodulators.

Almost all superhet receivers have tunable preselection filters to limit the input frequency range. As you will see in Chapter 6, these filters can negatively affect the sensitivity. Receiver designs in which sensitivity is more important than selectivity may use very wide tunable filters or fixed tuned preselectors for lower loss.

A tunable local oscillator (LO) allows the receiver to be tuned to different portions of its frequency range, which is offset from the receiver’s tuned frequency by a fixed amount. Common offsets are 455 kHz at HF, 10.7 MHz or 21.4 MHz in VHF and UHF, and 21.4 MHz, 60 MHz, or 160 MHz in microwave receivers. In modern receivers, the LO is usually a synthesizer that tunes rapidly and accurately to any frequency in its range with a single digital control input.

The LO signal and any signals passing through the input filter are both injected into a mixer. The mixer output then contains all of the following signals:

- All input signals from the filter
- The LO signal
- The sums and differences of those
- All harmonics of all input signals and their sums and differences

This mess is passed to an IF amplifier/filter that is at an offset frequency as previously discussed (e.g., 21.4 MHz). Since the IF filters are fixed tuned, they can have very sharp cutoffs, eliminating all of the mixing products except a replica of the signal coming through the input filter but centered at the IF. IF filtering also (usually) sets the predetection bandwidth of received signals. The receiver may
have several selectable IF bandwidths. The IF amplifier provides most or all of the gain to bring signals up to the level required for discrimination.

The discriminator (also called detector or demodulator) recovers the modulation from the received signal to reconstruct the original information signal that was input to the transmitter. The type of discriminator used depends on the type of modulation that has been applied to received signals. Many receivers have multiple selectable discriminator types (e.g., AM, FM, single sideband (SSB)).

This superhet approach can also move chunks of frequency spectrum around via a switched converter. In a typical application, a set of band-pass filters divides a wide input range into even segments, and each segment is then heterodyned to a single output band using the same approach used in the superheterodyne receiver, but with a switched local oscillator having only one frequency per band. This allows a relatively narrow receiver (e.g., a 2–4 GHz channelized receiver) to cover a much wider frequency range (e.g., 2–18 GHz).

### 4.3.4 Crystal Video Receiver

The crystal video receiver is shown in Figure 4-22. It has an AM (crystal) detector preceded by a band-pass filter and (usually) a preamplifier and followed by a video amplifier. It continuously covers its entire frequency range, detecting all signals present.

This means that the crystal video receiver has a 100% probability of detecting any AM signal it sees but cannot separate multiple overlapping signals. It is also severely limited in sensitivity since the detector operates on low-level received signals in the square law region. Superhet receivers provide better sensitivity because their detection takes place in the linear region after a significant amount of amplification.

Crystal video receivers are mainly useful for pulse signals, which have very short duty cycle and in many useful applications reach the receiver at strong signal levels. Although it provides good time of arrival, amplitude, and shape information about pulses, the crystal video receiver gives no information about the frequency of received pulses except that they are within the limits of its band-pass filter.

![Figure 4-22 Crystal video receiver](image)
4.3.5 IFM Receiver

Often used as a companion to crystal video receivers, instantaneous frequency measurement (IFM) receivers give very high-speed digital measurements of the frequency of received signals. They have approximately the same sensitivity as crystal video receivers and cover up to octave bandwidths.

Figure 4-23 shows the block diagram of a typical IFM receiver. A hard-limiting amplifier is required because the IFM discriminator is highly sensitive to received signal amplitude and can give accurate frequency measurements only when all input signals are at the same signal strength.

4.3.6 Compressive Receiver

The compressive receiver, shown in Figure 4-24, determines the frequency of all signals that are present in an octave-wide frequency range. It has good sensitivity, like a superheterodyne receiver, and can receive multiple simultaneous signals. It can give accurate measurements only when a single signal is present. This is also

![Figure 4-23 IFM receiver](image)

![Figure 4-24 Compressive receiver](image)
called a **micro-scan receiver** because it scans a wide IF amplifier very rapidly across a wide frequency range. The output of this IF is passed to a compressive filter, which has a delay proportional to frequency. The slope of the delay function (i.e., delay vs. frequency) is matched to the sweep of the local oscillator (i.e., its frequency vs. time). This causes each received signal to output to the detector for a prolonged period. Each detected signal is analyzed by a processor that outputs its frequency in digital form.

### 4.3.7 Digital Receiver

The digital receiver is in some ways the simplest type of receiver—at least in hardware. A received signal is merely digitized. Then a computer of some type performs all of the demodulation and filtering functions in software. The great advantage of the digital receiver comes in its flexibility. Since filters and demodulators are implemented in software, they can have virtually any parameters, including some that could not be achieved in hardware.

Digitized signals can be stored and sequentially subjected to different filtering and demodulation until the desired output quality is achieved. Stored, digitized frequency bands can be sequentially searched to detect types of signals that cannot even be observed until they have been subjected to some level of processing.

That was the good news. The bad news is that the state of the art in analog to digital converters is not adequate to digitize low-level RF signals in very wide bandwidths. Also, when predetection signals are digitized with enough resolution to provide good results, over any significant signal duration, they occupy vast amounts of memory.

Practical digital receivers require some compromises, as shown in Figure 4-25. First, a switchable translator is usually used to select a reasonable portion of RF spectrum to process. Earlier digitizers required that digitization be done in a **zero IF**. This meant that a translator was used to heterodyne the undiscriminated band down in frequency until its lower edge was near zero Hz. This is still done in some systems, but most new digitizers work directly at a VHF or UHF IF frequency.

![Figure 4-25 Digital receiver](image)

---

**Figure 4-25 Digital receiver**
This translation of any part of the receiver’s RF input brings the whole frequency band within the range of the ADC.

Some clever tricks (e.g., submultiple sampling) can be used under some circumstances to simplify or improve this process, but they will not be discussed here. Also, the upper frequency and throughput rate of ADCs and improved digital storage media are the subject of much serious research and development. Available performance changes almost daily.

Although there are limits to what they can do, digital receivers are already very useful in several important applications, and their applications are constantly expanding with the improvements in component state of the art.

4.3.8 Receiving Systems

Receivers are often organized into receiving systems including preamplifiers, signal distribution networks, and multiple receivers of the same or different types. Figure 4-26 shows a typical receiving system. It is important to note that the performance of each individual receiver in the communication link is significantly impacted by the characteristics of the hardware in the signal path between the antenna and the receiver’s input connector. There are often practical reasons to have receivers far separated from the antenna location. If so, the cable losses from the antenna to the preamplifier and from the preamplifier to the receivers can cause significant loss of link performance if the system is not properly designed.

Chapter 6 discusses receiver system sensitivity calculation techniques, including the effects of preamplifier specifications, receiver specifications, and system losses before and after the preamplifier.

![Figure 4-26 Receiver system](image)
4.4 Antennas

Antennas convert signal power from the transmitter into electromagnetic waves—and then convert them back into signal power that can be processed by the receiver. We characterize antennas in terms of their gain and directivity. The gain is the amount that the antenna increases the signal strength of signals that it transmits or receives. Directivity is the quality of providing more gain in one direction than the antenna provides in other directions. These two qualities are interactive, since antenna gain is the result of concentrating a transmitted signal into a limited angular space or, conversely, focusing on a limited angular space to concentrate the received signal. At any given angle, the antenna gain can be either positive or negative (in dB). Two other important antenna qualities discussed in this section are efficiency and polarization.

4.4.1 Antenna Gain Pattern

An isotropic antenna is a theoretical antenna that would provide exactly equal gain in all (spherical) directions. The gain of such an antenna would be one (1), that is, zero dB. True isotropic antennas do not exist for two reasons: (1) they would have to have 100% efficiency (which no antennas built by humans achieve), and (2) they would have to have a signal cable attached to it and be mounted somehow, but anything near an antenna distorts its gain pattern.

Still, the isotropic antenna is an important concept. Antenna gain in any direction is typically defined relative to the gain that would be provided by an isotropic antenna. Also, the assumption that an isotropic antenna is placed at some critical point in space allows the development of formulas that greatly simplify propagation calculations.

Figure 4-27 shows a typical antenna gain pattern. This is a polar plot of antenna gain versus angle like that generated in antenna test chambers (called anechoic chambers because their walls absorb all signal energy reaching them). An antenna is mounted in the center of the chamber and rotated through 360° either
vertically or horizontally. Signals are transmitted to the antenna, and its output is recorded as a polar plot of received strength, which represents the relative gain of the antenna versus angle.

As shown in the figure, the boresight of the antenna is normally its point of maximum gain. Zero degrees of angle is defined at the boresight, and the signal strength is usually plotted against a logarithmic scale. So the plot shows dB of gain versus degrees from boresight. The vertical pattern is called the elevation pattern, and the horizontal pattern is called the azimuth pattern.

Several other values are shown in this figure. The 1 dB beamwidth (BW) is the angular spread between the two points on either side of the boresight where the antenna gain is 1 dB less than the boresight gain. It is important to note that this is the angle between the two 1 dB points, not the angle from the boresight to the 1 dB point. (Lots of people make that mistake.)

The 3 dB beamwidth is the angle between the half power points of the antenna beam—that is, the points at which the antenna gain is 3 dB less than the boresight gain. When people talk about antenna beamwidth without defining it with some number of dB, they normally mean the 3 dB beamwidth. The 10 dB beamwidth is, of course, the angle between the –10 dB gain points (relative to boresight gain).

The main beam is the primary lobe of the antenna gain pattern, for which the antenna is designed. The sidelobes (including the backlobe) are normally treated as undesirable features of the antenna and are defined in terms of their average or minimum gain difference below the boresight gain. For example, “The sidelobes are 25 dB down.”

An important antenna concept is the trade-off of gain versus beamwidth, which is logical since a directional antenna directionally concentrates the transmitted power. If there were no sidelobes and no losses in the antenna hardware, the antenna would be 100% efficient. The efficiency achieved (always less than 100%) is a function of the type of antenna, the quality of its construction, the way it is mounted, and the frequency range it is designed to cover. Efficiency of 55% is common for narrow frequency range antennas. Figure 4-28 is a graph of boresight gain versus 3 dB beamwidth for a 55% efficient antenna.

To use this figure, move right from the antenna gain to the line, then down to the 3 dB beamwidth appropriate to that amount of gain, or reverse the procedure to find the gain for a specified beamwidth. The example on the chart shows that a 55% efficient antenna with a 3 dB beamwidth of 2º will have 37.5 dB gain.

Antenna Efficiency
Antenna efficiency can be described as the amount of power emitted within a cone described by the 3 dB beamwidth divided by the total amount of power emitted by a transmitting antenna. The same concept can be applied to receiving antennas. For antennas that operate over a narrow bandwidth (less than 10%), efficiency can be as high as 55%. However, antennas operating over a wider frequency range will have less efficiency. An important example is an electronic warfare antenna operating over the frequency range from 2 to 18 GHz, which will have approximately 30% efficiency.
Another assumption in Figure 4-28 is that the antenna pattern has the same vertical and horizontal beamwidth. This is not always the case, but the following two convenient equations can be used to determine the approximate boresight gain versus the 3 dB beamwidth in two mutually perpendicular planes.

The first of these equations is for 55% efficient antennas (a widely assumed value for parabolic dishes):

$$G \text{ (not in dB)} = \frac{29,000}{\theta_1 \times \theta_2}$$

where $\theta_1$ and $\theta_2$ are the 3 dB beamwidths (in degrees) measured in any mutually orthogonal planes.

Remember that this is not the gain in dB. To determine the gain in dB, use the dB conversion technique described in Chapter 2 to get

$$G \text{ (dB)} = 10 \log \left( \frac{29,000}{\theta_1 \times \theta_2} \right)$$

The same formula for antennas with 60% efficiency (a commonly assumed value for horns) is

$$G \text{ (not in dB)} = \frac{31,000}{\theta_1 \times \theta_2}$$

where $\theta_1$ and $\theta_2$ are the 3 dB beamwidths (in degrees) measured in any mutually orthogonal planes.

$$G \text{ (dB)} = 10 \log \left( \frac{31,000}{\theta_1 \times \theta_2} \right)$$

Another important way to describe antennas is in terms of their effective area. This does not always correspond in any satisfying way with the physical size of
the antenna, although they are related. The effective area can be mathematically defined in terms of the antenna gain and the frequency of the signal being transmitted or received using

\[ A = 38.6 + G - 20 \log(F) \]

where \( A \) is the effective area (in dBsm), \( G \) is the boresight gain (in dB), and \( F \) is the frequency (in MHz).

Remember that dBsm is the ratio of the antenna area to one square meter, converted into dB form. It should also be noted that 38.6 (which is often rounded to 39 when 1 dB accuracy is enough) is a combination of various constants and conversion factors to allow the gain to be input in dB and the frequency to be input in MHz. Appendix A includes a derivation of the aforementioned formulas.

Effective antenna area in square meters is found by converting the value for area in dBsm back into linear form using the technique described in Chapter 2:

\[ A(m^2) = 10^{\frac{A(dBsm)}{10}} \]

Figure 4-29 is a nomograph (based on the previous formulas) that conveniently relates peak antenna gain, effective antenna area, and frequency. This figure allows you the flexibility to determine any one factor in this three-way relationship in terms of the other two.

The example shown in this figure is our old friend the isotropic antenna (which you will remember has unity, that is, 0 dB) gain. Drawing a line from 1 m\(^2\) on the area line through 0 dB on the gain line to the frequency scale shows that an isotropic antenna has an effective area of 1 m\(^2\) at approximately 85 MHz.

### 4.4.2 Polarization

The polarization of an electromagnetic wave refers to the orientation of its electrical and magnetic fields. The critical thing to understand about polarization is that the transmitting and receiving antennas must have the same polarization, or the receiving antenna experiences an additional polarization mismatch loss.

The polarization of the transmitting antenna determines the polarization of the transmitted signal. Most antennas have one of four principal polarizations:

- **Vertical (V)**
- **Horizontal (H)**
- **Right-hand circular (RHC)**
- **Left-hand circular (LHC)**

In addition to these four, antennas can be linearly polarized at any angle (45° is fairly common) or can have elliptical polarization (particularly when viewed from an angle off of the boresight).

Polarization, unlike some aspects of communication theory, has a satisfying physical logic to it in most cases. Vertically oriented antennas (e.g., a common whip antenna) have vertical polarization, while horizontally oriented antennas have horizontal polarization. Most (but not all) circularly polarized antennas are round
Also satisfying is what happens to signals when they reflect from flat surfaces. For example, a vertically polarized signal that reflects from an angled surface will come away with a linear polarization that is no longer vertical.

Figure 4-30 shows the polarization loss for various signal polarizations when they are received by antennas with matched or unmatched polarization.

In the real world, received signals often include many multipath components in addition to the primary signal (e.g., reflections of the same signal from buildings, trees, the ground, the antenna support structure). This gives the received signal a cross-polarization component, so the actual polarization loss may be less than shown. However, where there is clear line of sight between the transmitting and receiving antennas, the direct signal will usually be much larger than the multipath components, so these loss figures can be considered fairly accurate estimates.
Polarization loss is not always bad. For example, a receiving system could be designed with an RHC polarized antenna to significantly reduce interference from a strong LHC polarized signal at the same frequency as the desired signal. This is often done in systems to send control signals from Earth stations to satellites.

A circularly polarized antenna is often used in the design of systems to receive many different linearly polarized signals—vertical, horizontal, or anything in between. The uniform 3 dB polarization loss is experienced for all received signals, but that is preferable to the much greater loss that would be experienced for cross-polarized signals.

### 4.4.3 Types of Antennas

Table 4-5 is a summary of the most common types of antennas. For each type, it shows the general gain pattern and *typical* specifications. It is important to understand that individual antennas can be designed to meet a range of gain, bandwidth, and beamwidth parameters around the typical values listed and that these parameters trade off against each other. Each antenna type can be used as either a transmitting or receiving antenna and will have the same specifications and general configuration. However, the construction of high-power transmitting antennas is somewhat different to handle the increased currents.

Because they are more conceptually complex, the last two antenna types in the table (dishes and phased arrays) are described in more detail following the table.
### Table 4-5  Types of Antennas

<table>
<thead>
<tr>
<th>Antenna type</th>
<th>Pattern</th>
<th>Typical specifications</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Dipole</strong></td>
<td><img src="image" alt="Dipole Pattern" /></td>
<td><strong>Polarization</strong>: Aligned with element orientation  \n<strong>Beamwidth</strong>: 80°×360°  \n<strong>Gain</strong>: 2 dB  \n<strong>Bandwidth</strong>: 10%  \n<strong>Frequency range</strong>: HF through μw</td>
</tr>
<tr>
<td><strong>Whip</strong></td>
<td><img src="image" alt="Whip Pattern" /></td>
<td><strong>Polarization</strong>: Vertical  \n<strong>Beamwidth</strong>: 45°×360°  \n<strong>Gain</strong>: 0 dB  \n<strong>Bandwidth</strong>: 10%  \n<strong>Frequency range</strong>: HF through UHF</td>
</tr>
<tr>
<td><strong>Loop</strong></td>
<td><img src="image" alt="Loop Pattern" /></td>
<td><strong>Polarization</strong>: Horizontal  \n<strong>Beamwidth</strong>: 80°×360°  \n<strong>Gain</strong>: –2 dB  \n<strong>Bandwidth</strong>: 10%  \n<strong>Frequency range</strong>: HF through UHF</td>
</tr>
<tr>
<td><strong>Normal mode helix</strong></td>
<td><img src="image" alt="Helix Pattern" /></td>
<td><strong>Polarization</strong>: Horizontal  \n<strong>Beamwidth</strong>: 45°×360°  \n<strong>Gain</strong>: 0 dB  \n<strong>Bandwidth</strong>: 10%  \n<strong>Frequency range</strong>: HF through UHF</td>
</tr>
<tr>
<td><strong>Axial mode helix</strong></td>
<td><img src="image" alt="Axial Helix Pattern" /></td>
<td><strong>Polarization</strong>: Circular  \n<strong>Beamwidth</strong>: 50°×50°  \n<strong>Gain</strong>: 10 dB  \n<strong>Bandwidth</strong>: 70%  \n<strong>Frequency range</strong>: UHF through low μw</td>
</tr>
<tr>
<td><strong>Biconical</strong></td>
<td><img src="image" alt="Biconical Pattern" /></td>
<td><strong>Polarization</strong>: Vertical  \n<strong>Beamwidth</strong>: 20° to 100°×360°  \n<strong>Gain</strong>: 0 to 4 dB  \n<strong>Bandwidth</strong>: 4 to 1  \n<strong>Frequency range</strong>: UHF through mmw</td>
</tr>
</tbody>
</table>
Table 4-5 (Continued)

<table>
<thead>
<tr>
<th>Antenna type</th>
<th>Pattern</th>
<th>Typical specifications</th>
</tr>
</thead>
</table>
| Lindenblad               | EI, Az  | Polarization: Circular  
Beamwidth: 80° × 360°  
Gain: –1 dB  
Bandwidth: 2 to 1  
Frequency range: UHF through μw |
| Swastika                 | EI, Az  | Polarization: Horizontal  
Beamwidth: 80° × 360°  
Gain: –1 dB  
Bandwidth: 2 to 1  
Frequency range: UHF through μw |
| Yagi                     | EI, Az  | Polarization: Horizontal  
Beamwidth: 90° × 50°  
Gain: 5 to 15 dB  
Bandwidth: 5%  
Frequency range: VHF through UHF |
| Log periodic             | EI, Az  | Polarization: Vertical or Horizontal  
Beamwidth: 80° × 60°  
Gain: 6 to 8 dB  
Bandwidth: 10 to 1  
Frequency range: HF through μw |
| Cavity backed spiral     | EI & Az | Polarization: R & L Circular  
Beamwidth: 60° × 60°  
Gain: –15 dB (min freq) +3 dB (max freq)  
Bandwidth: 9 to 1  
Frequency range: μw |
| Conical spiral           | EI & Az | Polarization: Circular  
Beamwidth: 60° × 60°  
Gain: 5 to 8 dB  
Bandwidth: 4 to 1  
Frequency range: UHF through μw |
### Table 4-5 (Continued)

<table>
<thead>
<tr>
<th>Antenna type</th>
<th>Pattern</th>
<th>Typical specifications</th>
</tr>
</thead>
</table>
| 4 Arm conical spiral          | ![4 Arm conical spiral](image) | Polarization: Circular  
Beamwidth: $50^\circ \times 360^\circ$  
Gain: 0 dB  
Bandwidth: 4 to 1  
Frequency range: UHF through μw |
| Horn                          | ![Horn](image) | Polarization: Linear  
Beamwidth: $40^\circ \times 40^\circ$  
Gain: 5 to 10 dB  
Bandwidth: 4 to 1  
Frequency range: VHF through mmw |
| Horn with polarizer           | ![Horn with polarizer](image) | Polarization: Circular  
Beamwidth: $40^\circ \times 40^\circ$  
Gain: 5 to 10 dB  
Bandwidth: 3 to 1  
Frequency range: μw |
| Parabolic dish                | ![Parabolic dish](image) | Polarization:  
Depends on feed  
Beamwidth: 0.5 to 30°  
Gain: 10 to 55 dB  
Bandwidth:  
Depends on feed  
Frequency range: UHF to μw |
| Phased array                  | ![Phased array](image) | Polarization:  
Depends on elements  
Beamwidth: 0.5 to 30°  
Gain: 10 to 40 dB  
Bandwidth:  
Depends on elements  
Frequency range: VHF to μw |
Parabolic Dish Antennas

Parabolic dish antennas (commonly just dishes) reflect signals from a feed antenna located at the focus of the parabola as shown in Figure 4-31. The feed antenna is normally a horn, log periodic, or spiral antenna. Dish antennas have a great deal of flexibility in application because their gain and beamwidth are dependent on their size and the operating frequency, while their polarization and bandwidth are set by the feed.

The relationship between the gain of a parabolic reflector antenna (with 55% efficiency), its diameter, and the operating frequency can be calculated from

\[ G = -42.2 + 20 \log(D) + 20 \log(F) \]

where \( G \) is the antenna gain (in dB), \( D \) is the reflector diameter (in meters), \( F \) is the frequency (in MHz), and –42.2 combines several constants and conversion factors (see Appendix A) to allow input of diameter and frequency in these units.

Figure 4-32 is a nomograph (based on the previous equation) of gain versus size versus frequency for 55% efficient parabolic dish antennas. A line from the frequency to the antenna diameter shows the boresight gain at the center scale. In the example, a 1/2 m diameter antenna operating at 10 GHz is shown to have approximately 32 dB of gain. Only relatively narrow frequency range antennas (about 10% bandwidth) achieve 55% efficiency. Wider frequency coverage by an antenna will reduce its efficiency.

Table 4-6 shows the gain adjustments necessary to compensate for antenna efficiency factors other than 55%. To use this table, find the antenna gain from Figure 4-32 and then adjust the gain by adding or subtracting the number of dB indicated in the table.

You can make antenna parameter trade-offs from these equations and nomographs, but you have a secret weapon included with this book: an antenna and radio propagation slide rule. It is described in Section 4.5, along with complete instructions.
Phased Array Antennas

A **phased array antenna** is an array of several simple antennas connected to a transmitter or receiver through variable delay lines as shown in Figure 4-33. This drawing is for a receiving antenna but would work the same for a transmitting antenna. The delay lines are adjusted so that the signals from the antennas add in-phase (maximizing the combined signal power) when the signal arrives from the **beam direction**. This causes an effective gain in the desired direction. Signals arriving from other directions add out of phase, which significantly decreases the combined signal power.

If electrically controlled delay lines are used, the phased array beam is said to be **electronically steered**. This gives it the great advantage over any other type
of directional antenna that it can be instantly moved to any pointing angle without the delays caused by rotating a mechanical antenna structure.

Phased arrays can be constructed using any type of antennas but usually use simple dipoles or flat spiral antennas that can be built into a flat surface (etched or deposited) with great precision. This gives them very low relative thickness, a great advantage for some applications (e.g., mounting on an aircraft). In high-power transmitting phased arrays, horn antennas are often used. They can be built into a single structure, which allows the use of very thin material for applications in which weight is a significant problem.

The length of a phased array and the operating frequency determine the beamwidth. For example, a 3 dB beamwidth of 3° requires an array 20 wavelengths long; that is, at 6 GHz it must be $20 \times 5$ cm, or 1 m long. The spacing of the antennas in the array determine the angular range over which the antenna can be steered without degrading the antenna performance below the specified value. A good typical number for the efficiency of a relatively narrow bandwidth phased array is 30%.

Table 4-7 is an antenna selection guide to assist in the selection of antenna type from the various specifications required by an application.

### 4.5 Antenna Slide Rule

The slide rule supplied with this book supports two types of calculations: (1) antenna parameter trade-offs and (2) propagation losses. The antenna scales are described here, and the propagation loss scales are described in Chapter 5. For your convenience, instructions for use of all of the scales are repeated in Appendix G. Checking many slide rule readings against the background math shows that the indexing of the printing and the accuracy of moving the slide limits the accuracy of slide rule readings to about two-tenths of a dB. If you need more accuracy than that, please use the calculations given elsewhere in this book.
4.5.1 Antenna Scales

The antenna scales are all on side 1 of the slide rule, as shown in Figure 4-34. These scales are most accurate for parabolic antennas but also give useful approximations of performance for other types of antennas. They allow calculation of boresight gain and beamwidth from the diameter of the antenna, operating frequency and efficiency, and also gain reductions for antennas without perfectly parabolic surfaces.

There are three assumptions for these antenna scales:

- The reflector is round (i.e., the vertical and horizontal dimensions are equal).
- The reflector is a perfect parabolic section (see Figure 4-35 if it is not).
- The dish contains 90% of the feed energy.

4.5.2 Finding Gain and Beamwidth

First, move the slide so that the frequency is next to the antenna diameter on the top window of side 1. Figure 4-35 shows 4 GHz against 8 feet.

Now move down to the second window and read the boresight gain against the efficiency. Figure 4-36 shows that the gain is 37.4 dBi at 55% efficiency.

Now look at the beamwidth scale at the top of the second window on side 1. You can read the 3 dB and 10 dB beamwidths on this scale. Figure 4-37 shows that the 3 dB beamwidth is about 2.25° and the 10 dB beamwidth is about 4.1°.

Now set the frequency at the third window on side 1 and look at the bottom window on side 1. The gain as calculated before must be reduced by the number of dB next to the surface tolerance. Figure 4-38 shows that at 4 GHz, a surface tolerance of 0.2 in root mean square (rms) reduces the gain by 3 dB. Note that rms means that measurements were made systematically or randomly over the whole surface of the dish. At each point, the variation from a perfect parabola was measured. These error values were squared, averaged, and the square root taken.

Table 4-7 Antenna Selection Guide

<table>
<thead>
<tr>
<th>Angular Coverage</th>
<th>Polarization</th>
<th>Bandwidth</th>
<th>Antenna Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>360° Coverage</td>
<td>Linear</td>
<td>Narrow</td>
<td>Whip, dipole, or loop</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Wide</td>
<td>Biconical or swastiika</td>
</tr>
<tr>
<td>Circular</td>
<td></td>
<td>Narrow</td>
<td>Normal mode helix</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Wide</td>
<td>Lindenblad or 4-arm conical spiral</td>
</tr>
<tr>
<td>Directional</td>
<td>Linear</td>
<td>Narrow</td>
<td>Yagi, array with dipole elements, or dish with horn feed</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Wide</td>
<td>Log periodic, horn, or dish with log periodic feed</td>
</tr>
<tr>
<td></td>
<td>Circular</td>
<td>Narrow</td>
<td>Axial mode helix, horn with polarizer or dish with crossed dipole feed</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Wide</td>
<td>Cavity-backed spiral, conical spiral, or dish with spiral feed</td>
</tr>
</tbody>
</table>

56 Practical Communication Theory
Parabolic antenna calculations

Set frequency at antenna diameter.

Read beamwidth and gain at index below.

- Positions (Location relative to the peak on the main beam), not beamwidths.

Set frequency at arrow.

Read free-space attenuation at desired range in meters or kilometers.

Read gain reduction at surface tolerance.

**Figure 4-34 Side 1 of slide rule**
### Figure 4-35  Slide rule set to 4 GHz and 8 feet

<table>
<thead>
<tr>
<th>Frequency - GHz</th>
<th>Diameter - Feet</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.04</td>
<td>600</td>
</tr>
<tr>
<td>0.06</td>
<td>500</td>
</tr>
<tr>
<td>0.08</td>
<td>400</td>
</tr>
<tr>
<td>0.10</td>
<td>300</td>
</tr>
<tr>
<td>0.15</td>
<td>200</td>
</tr>
<tr>
<td>0.20</td>
<td>100</td>
</tr>
<tr>
<td>0.3</td>
<td>80</td>
</tr>
<tr>
<td>0.4</td>
<td>60</td>
</tr>
<tr>
<td>0.5</td>
<td>50</td>
</tr>
<tr>
<td>0.6</td>
<td>40</td>
</tr>
<tr>
<td>0.8</td>
<td>30</td>
</tr>
<tr>
<td>1.0</td>
<td>20</td>
</tr>
<tr>
<td>1.5</td>
<td>15</td>
</tr>
<tr>
<td>2.0</td>
<td>10</td>
</tr>
<tr>
<td>3.0</td>
<td>8</td>
</tr>
<tr>
<td>4.0</td>
<td>6</td>
</tr>
<tr>
<td>5.0</td>
<td>4</td>
</tr>
<tr>
<td>6.0</td>
<td>3</td>
</tr>
<tr>
<td>8.0</td>
<td>2</td>
</tr>
<tr>
<td>10</td>
<td>1 1/2</td>
</tr>
<tr>
<td>15</td>
<td>1 3/4</td>
</tr>
<tr>
<td>20</td>
<td>1/2</td>
</tr>
<tr>
<td>25</td>
<td>1/4</td>
</tr>
<tr>
<td>30</td>
<td>1/8</td>
</tr>
</tbody>
</table>

*1st sidelobe-max*
**Figure 4-36** Gain vs. efficiency

**Figure 4-37** 3 dB beamwidth, 10 dB beamwidth, 1st sidelobe max and 1st null

**Figure 4-38** Gain reduction vs. surface tolerance
4.6 The Ether Waves

The rest of the communication link is what happens between the transmitting and receiving antennas as the signals pass through the atmosphere (or space) between them. Chapter 5 contains formulas, nomographs, and slide rule scales describing the path losses for the following cases:

- Clear line of sight
- Transmission near the ground or water
- Transmission over or near ridge lines

When signals must pass through rain or fog, the communication link will still function, but there will be additional losses, as described in Chapter 7.
Chapter 5

The Link Equation

The link equation described in this chapter calculates received signal power as a function of the various link parameters described in Chapter 4. It is also sometimes called the one-way link equation to distinguish it from radar equations that deal with round-trip propagation.

It is a little misleading to talk about the link equation as though it were a single equation; several equations are actually used to answer the real-world questions you want answered:

- How much transmitter power is required to transmit over a given range?
- How much receiver sensitivity is required to adequately receive a given signal at a given range?
- How much antenna gain (transmit or receive) is required?
- What is the effective range of a transmitter with a particular output power?

Like most real-world problems, the answer to each of these questions is, that depends. Each of these questions is readily answered using different forms of the link equation with applicable values plugged in.

Figure 5-1 shows an overview of what happens to a signal as it progresses through the communication link. The signal:

- Originates in a transmitter that outputs a transmitted power level
- Is radiated by an antenna which provides gain
- Is reduced in signal strength by link losses
- Is received by an antenna that provides gain
- And arrives at a receiver with a received power level

The whole purpose of the link equation is to calculate the signal strength arriving at the receiver, given all of the other factors. Thus, in its simplest form (with all values in dB forms), the link equation is

\[
\text{Transmitter power} + \text{Transmitter antenna gain} - \text{Link losses} + \text{Receiving antenna gain} = \text{Received power}
\]
The most commonly used units are

- Transmitted power in dBm
- Link losses in dB
- Antenna gains in dB
- Received power in dBm

Remember that the antenna gain (in dB) can be either positive or negative and is also a function of the angle at which the signals arrive or leave (i.e., relative antenna orientation). The gain used in this equation (for both transmitting and receiving antennas) is the gain in the direction of the other antenna whether or not the boresights of the antennas are aligned to each other. Throughout this book, losses are considered to be positive numbers, so they can be subtracted in the various link equation forms.

**Example**

Transmitter power = 10 watts (= +40 dBm)
Transmitting antenna gain = 6 dB
Link losses = 100 dB
Receiving antenna gain = −2 dB

Signal strength received by the receiver = −56 dBm

There will be more about using this equation to solve those practical problems after discussing the causes and calculation of link losses.

![Figure 5-1 Signal strength variations through the link](image)
5.1 Link Losses

In this chapter, we deal with three types of propagation loss models; each is assumed to be in good weather. Note that transmission in rain and fog is covered in Chapter 7. The three propagation models are as follows:

- A free-space link: there are no obstructions between the two antennas (including the curvature of the earth), and the signal path stays a wavelength or two away from the surface.
- A two-ray link: there are no obstructions between the two antennas, but both are near the ground or water.
- A knife-edge diffraction link: the line between the transmitting and receiving antennas either comes close to a terrain obstruction such as a ridge line or is actually below the obstruction so that the signal must bend over the obstruction to reach the receiving antenna.

In all of these models, there is an additional atmospheric loss that is a function of frequency and distance through the atmosphere.

5.1.1 Free Space

Free-space loss is caused by the dispersion of signal energy as it radiates away from the transmitting antenna. It is also called line of sight loss, spreading loss, or range squared loss. The free-space propagation model applies when any one of the following conditions applies: narrow antennas (that preclude significant ground reflection); high frequency (we’ll put some numbers on this later); or the antennas are far from the ground (again, some numbers later).

From a perfect isotropic antenna, radio waves would spread out spherically, like someone was blowing up a balloon. To continue the analogy, the balloon gets thinner as it expands because the same amount of rubber is distributed over the expanding surface area of the sphere. The transmitted signal energy is likewise distributed over an expanding spherical surface.

Leaving the analogy before the balloon pops, the total signal energy remains the same at any distance from the transmitter but is distributed over the surface of a huge sphere of radius equal to that of distance (d). The surface area of that sphere is $4\pi d^2$. If we had a receiving antenna that could capture the whole sphere, there would be no spreading loss. However, any practical antenna will capture only a small part of the sphere as shown (in two dimensions) in Figure 5-2. The effective antenna area (discussed in Section 4.4.1) and the radius of the sphere (i.e., the distance from the transmitter) determine the fraction of the sphere’s surface captured by the antenna.

In the formula for free-space loss presented here, it is assumed that both the transmitting and receiving antennas are isotropic (with 0 dB gain). This is a great convenience since it allows us to add the gains (in dB) of the transmitting and receiving antennas as separate numbers when determining received signal strength.
From the ratio of the surface area of the expanding sphere and the effective area of an isotropic receiving antenna (which varies with frequency) we get a well-behaved formula for the spreading loss between two isotropic antennas:

$$ L = \frac{[(4\pi)^2d^2]/{\lambda}^2} $$

where $L$ is the free space loss ratio, $d$ is the link distance in meters, and $\lambda$ is the transmitted signal wavelength in meters.

In dB form, this equation becomes

$$ L_s = K + 20 \log(F) + 20 \log(d) $$

The constant ($K$) is another of those contrived numbers that include several constants and unit conversion factors to allow convenient input and output units. If you input the distance in kilometers and the frequency in MHz, the constant will be 32.44 (commonly rounded to 32 for 1 dB accurate calculations). The spreading loss is then defined by

$$ L_{FS} = 32 + 20 \log(F) + 20 \log(d) $$

where $L_{FS}$ is spreading loss in dB, $F$ is frequency in MHz, and $d$ is distance in km.

We will be using this formula often in the later parts of this chapter. For your convenience, this same equation is presented in Appendix E for distance in statute or nautical miles; the only difference is that each has a different constant. The formula is derived in Appendix A.

It is important not to fall in love with the expanding balloon analogy, which only literally applies if there is a truly isotropic transmitting antenna. The same formula applies equally well for transmission between any two antennas that have unobstructed line of sight to each other.

The $L_{FS}$ equation tells what the loss would have been if both were truly isotropic, which allows the antenna gains to be added separately in link equations.
When a dB or two of accuracy is plenty, you may find it more convenient to use the spreading loss nomograph shown in Figure 5-3. Draw a line from the frequency (in GHz) on the left-hand scale to the distance (in km) on the right-hand scale. The free-space loss in dB is the point at which the line crosses the center scale. In the example shown on the nomograph, the spreading loss for a 1 GHz signal at 20 km is about 118 dB. (Remember that the constant in the $L_{FS}$ equation was rounded down from 32.44 to 32.) This same nomograph is presented for distance in statute or nautical miles in Appendix E.

An additional way to calculate the spreading (or free-space) loss is by using the free-space attenuation scales on side 1 of your antenna and propagation slide rule (Figure 5-4; it is called free-space loss on the slide rule). Figure 5-4 shows the front (#1) side of the slide rule with the line of sight loss calculation area of the rule outlined.

Figure 5-5 shows just the free-space calculation portion of the rule. To use the slide rule for $L_{FS}$ calculation, move the slide so that the transmit frequency is by the arrow at point A on Figure 5-5.

In the example shown in this figure, the frequency is 300 MHz (0.3 GHz), the link distance is 25 km, and the free-space attenuation is 110 dB.

A shorter range scale for free-space loss is also provided on the slide rule. As shown in Figure 5-6, set the frequency at point A and read the free space loss against the link range in meters on the upper free-space attenuation scale. For this example, the frequency is still 300 MHz. Take the range as 25 m (point C on the slide rule) and read the free-space attenuation at just less than 50 dB.
**Parabolic antenna calculations**

Set frequency at antenna diameter.

Read beamwidth and gain at index below.

*Positions (location relative to the peak on the main beam), not beamwidths.*

---

**Figure 5-4** Front of antenna/propagation slide with line of sight attenuation scales highlighted.
5.1.2 Two-Ray Loss

When the transmitting and receiving antennas are close to a single dominant reflecting surface (i.e., the ground or water) and the antenna patterns are wide enough to allow significant illumination of that surface, the two-ray propagation model must be considered. As we will see, the transmitted frequency and the actual antenna heights determine whether the two-ray or line of sight propagation model applies.

Two-ray propagation is also called $40 \log d$ or $d^4$ attenuation because the loss varies with the fourth power of the link distance. The dominant loss in two-ray propagation is the phase cancellation of the direct wave by the signal reflected from the ground or water as shown in Figure 5-7. The amount of attenuation depends on the link distance and the height of the transmitting and receiving antennas above the ground or water. This section provides three ways to calculate two-ray loss:

- A formula (in linear or dB form)
- A nomograph
- The slide rule provided with this book

**Formula**

You will note that (unlike line of sight attenuation) there is no frequency term in the two-ray loss expression. In nonlogarithmic form, the two-ray loss is

$$L = \frac{d^4}{(h_T^2 \times h_R^2)}$$
where $d$ is the link distance, $h_T$ is the transmitting antenna height, $h_R$ is the receiving antenna height, and the link distance and antenna heights are all in the same units.

The dB formula for the two-ray propagation loss is

$$L = 120 + 40 \log(d) - 20 \log(h_T) - 20 \log(h_R)$$

where $d$ is the link distance in km, $h_T$ is the transmitting antenna height in meters, and $h_R$ is the receiving antenna height in meters.

**Nomograph**

Figure 5-8 gives a nomograph for the calculation of two-ray loss. To use this nomograph, first draw a line between the transmitting and receiving antenna heights.

---

*Figure 5-7  Direct and reflected rays close to the ground*

*Figure 5-8  Two-ray propagation loss nomograph*
Then draw a line from the point at which the first line crosses the index line through the path length to the propagation loss line. In the example, two 10 m high antennas are 30 km apart, and the attenuation is a little less than 140 dB. If you calculate the loss from either of the previous formulas, you will find that the actual value is 139 dB. Note that the two antenna heights need not be equal for this nomograph to work.

**Slide Rule**

Figure 5-9 shows the back side (2) of the slide rule with the scales for calculating two-ray loss highlighted. Figure 5-10 is a close-up of those scales. First move the slide so that the transmit antenna height (in meters) is aligned with the link distance in kilometers. Then read the attenuation in dB at the receiving antenna height in meters. For the example shown in the figure, the link distance is 20 km. The transmitting antenna height (2 m) is set next to 20 km at point A. Then the link loss is read at the receiving antenna height (30 m) at point B. Note that the link attenuation is shown to be about 136.5 dB. You need to be a little careful when interpolating between numbered points on the attenuation scale; the attenuation increases as you move left on the scale, so the attenuation is 136.5 dB, not 143.5 dB.
Minimum Antenna Height

Figure 5-11 shows minimum antenna height for two-ray propagation calculations versus transmission frequency. There are five lines on the graph for:

- transmission over sea water
- vertically polarized transmission over good soil
- vertically polarized transmission over poor soil
- horizontally polarized transmission over poor soil
- horizontally polarized transmission over good soil

*Good soil* provides a good ground plane. If either antenna height is less than the minimum shown by the appropriate line in this graph, the minimum antenna height should be substituted for the actual antenna height before completing the two-ray attenuation calculation. Please note that if one antenna is actually at ground level this chart is highly suspect. When an antenna is less than a half wavelength from the ground, its characteristics change in ways that are hard to predict and are considered beyond the scope of this book. To use this chart, start with the signal frequency. Move up to the appropriate soil and polarization line, and then move left to the minimum antenna height.

5.1.3 Fresnel Zone

Signals propagated near the ground or water with a clear path between the transmit and receive antennas can experience either free-space line of sight or two-ray
propagation loss depending on the antenna heights and the transmission frequency. The Fresnel zone distance is the distance from the transmitter at which the phase cancellation becomes dominant over the spreading loss. As shown in Figure 5-12, if the receiver is less than the Fresnel zone distance from the transmitter free-space propagation takes place. If the receiver is farther than the Fresnel zone distance from the transmitter two-ray propagation applies. In either case, the applicable propagation applies over the whole link distance.

This section provides two ways to determine the Fresnel zone distance:

- A formula (in two forms, neither in dB)
- The slide rule provided with this book

**Formula**
The Fresnel zone distance is calculated from

\[ FZ = 4\pi \frac{h_T h_R}{\lambda} \]

where \( FZ \) is the Fresnel zone distance in meters, \( h_T \) is the transmitting antenna height in meters, \( h_R \) is the receiving antenna height in meters, and \( \lambda \) is the transmission wavelength in meters.

Note that several different formulas for Fresnel zone are found in literature. This one is chosen because it yields the distance at which line of sight and two-ray attenuation are equal. A more convenient form of this equation is

\[ FZ = \left(\frac{h_T \times h_R \times F}{24,000}\right) \]

where \( FZ \) is the Fresnel zone distance in km, \( h_T \) is the transmitting antenna height in meters, \( h_R \) is the receiving antenna height in meters, and \( F \) is the transmission frequency in MHz.

**Slide Rule**
Figure 5-13 shows the back side (2) of the slide rule with the Fresnel zone calculation scales highlighted. Figure 5-14 shows a close-up of those scales.
To calculate the Fresnel zone distance, first move the slide so that the transmit antenna height in meters is aligned with the receiving antenna height in meters on the upper scale. Then read the Fresnel zone distance at the frequency in MHz on the lower scale. For the example shown in the figure, move the slide so that the transmit antenna height (2 m) is lined up with the receiving antenna height (30 m) at point A. Then read the Fresnel zone distance in kilometers at the frequency (200 MHz) at point B. The Fresnel zone distance is found to be 0.5 km, so if the link is shorter than 500 m free-space attenuation applies; if the link is longer than 500 m two-ray attenuation applies.

Note that when both the transmit and receive antennas are low the Fresnel zone distance may be off the end of the scale. But you will note that it is off the low end of the scale (i.e., much less than 0.1 km). This means that the Fresnel zone distance is much less than the link distance, so you know that the propagation mode is two ray.

**Complex Reflection Environment**

In locations with very complex reflections (e.g., when transmitting down a valley as shown in Figure 5-15), it is suggested in the literature that the free space propagation loss model will give a more accurate answer than the two-ray propagation model.
5.1.4 Knife-Edge Diffraction

Non–line of sight propagation over a mountain or ridge line is usually estimated as though it were propagation over a knife edge. This is a very common practice, and many electronic warfare professionals report that the actual losses experienced in terrain closely approximate those estimated by equivalent knife edge diffraction (KED) estimation. In this section only a nomograph for the calculation of KED loss is provided. The underlying math is really ugly.

The KED attenuation is added to the free-space loss as it would be if the knife edge were not present. Note that the free-space loss rather than the two-ray loss applies when a knife edge (or equivalent) is present.

The geometry of the link over a knife edge is shown in Figure 5-16. $H$ is the distance from the top of the knife edge to the line of sight as though the knife edge were not present. The distance from the transmitter to the knife edge is called $d_1$, and the distance from the knife edge to the receiver is called $d_2$. For KED to take place, $d_2$ must be at least equal to $d_1$. If the receiver is closer to the knife edge than the transmitter, it is in a blind zone in which only tropospheric scattering (with significant loss) provides link connection.

The knife edge causes loss even if the line of sight passes above the peak—unless the line of sight path passes several wavelengths above. As shown in Figure 5-17, $H$ can be either the distance above or below the knife edge.
Figure 5-18 is a KED calculation nomograph. The left-hand scale is a distance value \( d \) that is calculated by

\[
d = \frac{\sqrt{2}}{1 + \frac{d_1}{d_2}} d_1
\]

Table 5-1 shows some calculated values of \( d \).

If you skip this step and just set \( d = d_1 \), the accuracy of the KED attenuation estimation will be reduced only by about 1.5 dB.

Returning to Figure 5-18, the line from \( d \) (in km) passes through the value of \( H \) (in meters). At this point, we don’t care whether \( H \) is the distance above or below the knife edge. Extend this line to the center index line.
Another line passes from the intersection of the first line with the center index through the transmission frequency (in MHz) to the right-hand scale, which gives the KED attenuation. At this point, we identify whether \( H \) was above or below the knife edge. If \( H \) is the distance above the knife edge, the KED attenuation is read on the left-hand scale. If \( H \) is the distance below the knife edge, the (greater) KED attenuation is read on the right-hand scale.

Consider an example drawn onto the nomograph, where \( d_1 \) is 10 km, \( d_2 \) is 24.1 km, and the line of sight path passes 45 m below the knife edge.

In the example, \( d \) is 10 km (from Table 5-2), and \( H \) is 45 m. The frequency is 150 MHz. If the line of sight path were 45 m above the knife edge, the KED attenuation would have been 2 dB. However, since the line of sight path is below the knife edge, the KED attenuation is 10 dB.

The total link loss is then the line of sight loss without the knife edge + the KED attenuation.

\[
\text{Free-space loss} = 32.44 + 20 \log(d_1 + d_2) + 20 \log(\text{frequency in MHz}) \\
= 32.44 + 20 \log(34.1) + 20 \log(150) \\
= 32.44 + 30.66 + 43.52 = \text{approximately 106.6 dB}
\]

So the total link loss is \( 106.6 + 10 = 116.6 \text{ dB} \).

5.1.5 Atmospheric Loss

Atmospheric loss is caused by the absorption of radio energy (mainly by oxygen and water vapor) as signals pass through the earth’s atmosphere. The combined losses per kilometer are shown as a function of frequency in Figure 5-19. Note that this curve is for attenuation horizontally through the troposphere near the earth’s surface.

To use this chart, enter at the transmission frequency along the bottom of the chart, move straight up to the curve and read the loss per unit distance on the left scale. Assuming that the whole transmission path is in the atmosphere, the total atmospheric loss is the loss per unit distance multiplied by the link distance. For example, at 47 GHz the atmospheric loss is about 0.4 dB per kilometer, so a 10 km link at 47 GHz would have 4 dB of atmospheric loss.

Notice that there is a very strong peak in atmospheric attenuation at about 60 GHz, which would make this a very poor choice for an earth transmission link.
but excellent for transmitting from satellite to satellite if you didn’t want anyone on
the earth to listen in.

Another thing to notice about this chart is that the atmospheric attenuation gets
very small at low frequencies. Below microwave frequencies, it is usually
acceptable to ignore the atmospheric attenuation as negligible, particularly when
making calculations to 1 dB accuracy.

Equivalent curves are given in Appendix E for atmospheric attenuation per
statute mile and nautical mile.

Figure 5-20 shows a family of curves for the atmospheric attenuation through
the Earth’s whole atmosphere for satellite to ground links, as a function of frequency
and the elevation angle of the satellite above the horizon at the ground station. There
is naturally much higher attenuation at low elevation angles because much more of
the transmission path is within the atmosphere.

5.2 Signal Strength at Various Points in the
Communication Link

In the design or evaluation of a communication link, it is necessary to pin down the
signal strength at various points in terms of the link variables (transmitter power,
antenna gains, and link losses). This section deals with the link points commonly
considered. In each case, an equation for the signal strength at that point in the
link will be given in terms of the other defined signal strengths, gains, and losses in the link.

5.2.1 Effective Radiated Power

The effective radiated power (ERP) is the signal power radiated from the transmitting antenna in the direction of the receiving antenna. It is typically expressed in dBm. It is the sum of the transmitter power (in dBm) and the applicable antenna gain (in dB).

\[ \text{ERP} = P_T + G_T \]

Where:
- \( P_T \) = Transmitter power (dBm)
- \( G_T \) = Transmit antenna gain (dB)

*Figure 5-20  Attenuation through full atmosphere*
As shown in Figure 5-21, the ERP is very much a function of the link geometry. Assuming a transmitter power of 10 W (+40 dBm): if the receiver is at point A, the ERP is +50 dBm; if it is at point B, the ERP is only +41 dBm. However, if someone talks about ERP without mentioning the geometry, he or she means the ERP at the peak gain of the antenna.

Defining ERP in dBm is very useful and will give accurate results. However, it must be noted that we are taking some literary license in describing ERP in power units (dBm) since it is actually out in the ether waves between the two antennas. In fact, the signal travels between the two antennas in the form of an electromagnetic wave, which is more accurately defined in terms of its electrical field density (in microvolts per meter). Power (in milliwatts or dBm) is technically defined only inside a circuit.

The artifice used is to describe the electromagnetic wave’s field density in terms of the power that would be produced at the output of an isotropic (i.e., unity gain) antenna placed at the point we are considering. This approach immensely simplifies the bookkeeping and is accurate enough for all but the most hard-core technical types.

In the case of ERP, this imaginary isotropic antenna would be placed right at the antenna, so close that there would be no link loss (but ignoring the effects of being in the antenna’s near field, which would distort the results).

In case you need to express ERP in field density units, you can convert back and forth between microvolts per meter and dBm using the following formulas (which are derived in Appendix B):

To convert from $\mu$V/m to dBm

$$ P = -77 + 20 \log(E) - 20 \log(F) $$

where $P$ is the signal strength in dBm, $E$ is the field strength in $\mu$V/m, and $F$ is the frequency in MHz.
To convert from dBm to \( \mu \text{V/m} \)

\[
E = 10^{(P+77+20 \log(F))/20}
\]

Using the antilogn convention, this equation is

\[
E = \text{Antilogn}([P + 77 + 20 \log(F)]/20)
\]

### 5.2.2 Signal Arriving at Antenna

The strength of signals arriving at the antenna \( P_A \) is a useful consideration because it is often necessary to evaluate trade-offs of different types of antennas. It can also be important to determine the effect of an antenna gain pattern in the rejection of interfering signals.

The magnitude of the signal arriving at the receiving antenna is the effective radiated power (ERP) in the direction of the receiving antenna reduced by all link losses. This includes the spreading or two-ray loss and the atmospheric loss for line of sight links in good weather and also the knife edge refraction and the losses described in Chapter 7 for more challenging propagation situations.

\[
P_A = \text{ERP} - L
\]

Where:

- \( P_A \) = Signal power arriving at antenna (dBm)
- \( \text{ERP} \) = Effective radiated power (dBm)
- \( L \) = All link losses (in dB)

The link losses (for good weather, line of sight links) include propagation loss and atmospheric loss. The propagation loss can be from any of the propagation losses described in Section 5.1. For example, assume that the correct propagation loss is free space as described in Section 5.1.1. The expression for the power arriving at the antenna in good weather would then be

\[
P_A = \text{ERP} - 32 - 20 \log(F) - 20 \log(d) - L_{Atm}
\]

where \( P_A \) is the signal strength at receiving antenna (dBm), \( \text{ERP} \) is the effective radiated power (dBm), \( F \) is the frequency (MHz), \( d \) is the distance (km), and \( L_{Atm} \) is the atmospheric loss (dB).

For example, if the signal frequency is 100 MHz, the ERP is +50 dBm, and the receiver is 50 km from the transmitter with clear line of sight and good weather, the signal power arriving at the receiving antenna would be

\[
+50 \text{ dBm} - 32 \text{ dB} - 20 \log(100) \text{ dB} - 20 \log(50) \text{ dB} = +50 \text{ dBm} - 32 \text{ dB} - 20(2) \text{ dB} - 20(1.7) \text{ dB} = +50 \text{ dBm} - 32 \text{ dB} - 40 \text{ dB} - 34 \text{ dB} = -56 \text{ dBm}
\]

Note that atmospheric attenuation is negligible at 100 MHz.
Like ERP, the signal strength arriving at the receiving antenna can be easily converted from dBm to $\mu$V/m using the formulas given in Section 5.2.1. This may be quite useful since the sensitivity of some types of receiving systems with integral antennas (e.g., direction-finding systems) is sometimes specified as field strength in $\mu$V/m.

### 5.2.3 Received Power

You will notice that received power is the answer in the link equation as defined at the beginning of this chapter. It is the signal power input to the receiver from the receiving antenna and is equal to the signal strength arriving at the antenna plus the effective receiving antenna gain.

$$ P_R = ERP - L + G_R $$

Where:
- $P_R$ = Signal power into receiver (dBm)
- $ERP$ = Effective radiated power (dBm)
- $L$ = All link losses (dB)
- $G_R$ = Receiving antenna gain (dB)

It is normally assumed that the receiving antenna gain value is its peak gain, but this is not always the case. For example, when an interfering signal is received from a direction away from the intended transmitter, its received power is calculated using the antenna gain in the direction of the interfering transmitter. Another important case is when a narrow beam receiving antenna cannot be perfectly aligned. In this case, $G_R$ may be the gain at the maximum anticipated misalignment angle.

There are often significant cable runs between the receiving antenna and the receiver, and there may also be preamplifiers involved, so there is an important bookkeeping issue here. You need to be very clear about exactly where the received power is defined. In this book, it will always be defined at the output of the receiving antenna, since that is the point at which the antenna’s gain is defined. This means that any preamplifiers or cable runs are considered part of the receiver (or receiving system).

When the previous equation for received signal power is combined with the expression for free-space loss in good weather, it yields

$$ P_R = ERP - 32 - 20 \log(F) - 20 \log(d) - L_{Atm} + G_R $$

where $P_R$ is the signal strength into receiver in dBm, $ERP$ is the effective radiated power in dBm, $F$ is the frequency in MHz, $d$ is the distance in km, $L_{Atm}$ is the atmospheric loss in dB, and $G_R$ is the receiving antenna gain in dB.
For example, if the signal frequency is 100 MHz (so $L_{Atm}$ is negligible), the ERP is $+50$ dBm, the receiver is 50 km from the transmitter within line of sight, and the receiving antenna gain is 3 dB (in the direction toward the transmitter) the received power would be

$$+50 \text{ dBm} - 32 \text{ dB} - 20 \log(100) \text{ dB} - 20 \log(50) \text{ dB} + 3 \text{ dB}$$

$$= +50 \text{ dBm} - 32 \text{ dB} - 20(2) \text{ dB} - 20(1.7) \text{ dB} + 3 \text{ dB}$$

$$= +50 \text{ dBm} - 32 \text{ dB} - 40 \text{ dB} - 34 \text{ dB} + 3 \text{ dB} = -53 \text{ dBm}$$

### 5.3 Link Design Parameters

This section describes several important link design parameters in terms of the link variables that control them and the signal levels to which they relate.

#### 5.3.1 Required Margin

Because it is an imperfect world, things go wrong—particularly in radio communication links. It might rain, antennas may not stay perfectly aligned, there may be fading caused by reflections of our desired signal (from stationary or moving objects), or there may be unexpected external sources of noise or interference. To make the link dependable, more than the minimum acceptable signal level must be provided for the receiver.

\[
M = P_R - S
\]

Where:
- $M$ = Link margin (dB)
- $P_R$ = Received power (dBm)
- $S$ = Receiving system sensitivity (dBm)

The minimum signal level that a receiver can receive and still do its job is called its sensitivity. Sensitivity is a signal power level and is usually expressed in dBm. (You will learn to calculate the sensitivity in Chapter 6.) The amount by which the received power level exceeds the sensitivity is called the link margin. Margin is expressed in dB and can be achieved either by increasing the arriving signal strength or by improving the sensitivity. The tools available for providing margin depend on what parts of the link and the link geometry are under your control. They could include:

- Increasing the transmitter power
- Increasing the transmitting antenna gain
- Increasing the receiving antenna gain
- Decreasing the operating range
- Improving the receiver sensitivity
The amount of margin required depends on the link’s operating situation and on how important it is that the link be reliable. A general rule of thumb is that you should provide a margin equal to the inverse of the average dropout time ratio you can tolerate—converted to dB. Examples are:

- For 10% link dropout time, provide 10 dB margin ($1/0.1 = 10$ $10 \log(10) = 10$ dB)
- For 1% link dropout time provide 20 dB margin ($1/0.01 = 100$ $10 \log(100) = 20$ dB)
- For 0.1% link dropout time provide 30 dB margin ($1/0.001 = 1000$ $10 \log(1000) = 30$ dB)

### 5.3.2 Required Sensitivity

The sensitivity of a receiver is a specification of the minimum signal strength it can receive and still reproduce the transmitted information signal with adequate quality.

If you must provide a receiving system to receive signals from an existing transmitter, at some specified range the only variables you have to work with are the receiving antenna gain and the receiver sensitivity. Assuming for a moment that you must also work with an existing receiving antenna, you can calculate the required receiver sensitivity as follows:

$$S_{Rqd} = ERP - L + G_R - M$$

Where:

- $S_{Rqd}$ = Required sensitivity (dBm)
- $ERP$ = Effective radiated power (dBm)
- $L$ = All link losses (dB)
- $G_R$ = Receiving antenna gain (dB)
- $M$ = Required link margin (dB)

Again, if free-space propagation applies, this can be combined with the expression for free-space loss given in Section 5.1.1 to make the following very useful formula.

**For Good Weather Line of Sight Links**

$$S_{Rqd} = ERP - 32 - 20 \log(F) - 20 \log(d) - L_{Atm} + G_R - M$$

where:

- $S_{Rqd}$ = Required sensitivity in dBm
- $ERP$ = Effective radiated power in dBm
- $F$ = Operating frequency in MHz
- $d$ = Distance from transmitter in km
For example, if the signal frequency is 100 MHz (so $L_{Atm}$ is negligible), the ERP is $+50$ dBm, the receiver is 50 km from the transmitter within line of sight, the receiving antenna gain is 3 dB (in the direction toward the transmitter), and you must have 20 dB of performance margin the required receiver sensitivity (in dBm) would be:

$$+50 \text{ dBm} - 32 \text{ dB} - 20 \log(100) \text{ dB} - 20 \log(50) \text{ dB} + 3 \text{ dB} - 20 \text{ dB}$$

$$= +50 \text{ dBm} - 32 \text{ dB} - 20(2) \text{ dB} - 20(1.7) \text{ dB} + 3 \text{ dB} - 20 \text{ dB}$$

$$= +50 \text{ dBm} - 32 \text{ dB} - 40 \text{ dB} - 34 \text{ dB} + 3 \text{ dB} - 20 \text{ dB} = -73 \text{ dBm}$$

### 5.3.3 Effective Range

The effective range of a communication link is the maximum distance between the transmitting and receiving antennas for which the link can still do its job. You will often read an advertisement for a transmitter or a receiver with a *range of five miles*. This may or may not be true depending on a number of assumptions. Accurately specifying effective range requires the specification of every element of the link.

The only effect that operating range has on a line of sight link is the amount of link loss it causes. Thus, the effective range is the transmitter to receiver distance at which the maximum acceptable link loss occurs. This makes the received signal equal to the receiver sensitivity plus the required margin. The total link loss is customarily considered in two parts: propagation loss and atmospheric loss.

$$L = L_P + L_{Atm}$$

Both are dependent on the operating range. Figure 5-18 shows atmospheric loss per kilometer (through the atmosphere). It is a nonlinear function that does not readily lend itself to mathematical characterization. Propagation loss, however, is a well-behaved function of the expressions presented in Section 5.1.1 (for free-space loss) or 5.1.2 (for two-ray loss).

Since the atmospheric loss is insignificant at lower frequencies and is significantly less than the spreading loss for almost all links, the normal approach to determining effective range is to do the following:

1. Determine the acceptable total link loss.
2. Estimate the approximate atmospheric loss to a dB or so and subtract this amount from the total link loss (at UHF and below the estimate is typically zero).
3. Calculate the range at which the spreading loss causes the received signal to be equal to the sensitivity + the required margin. (This is the effective range.)
4. Check the atmospheric loss at that calculated range. If it is significantly different from the original atmospheric loss estimate, go back to step 2 with the new estimate and calculate a new effective range.
Yes, you can also set up an equation with the two types of losses vs. range and solve for range. However, it’s an ugly equation because the atmospheric loss varies with the range and a noncalculable function of frequency, whereas the propagation loss varies with range and frequency squared (for free space) or range and antenna heights (for two ray). Almost nobody does it that way unless they have a lot of calculations to do or can run them on a computer.

Now, let’s focus on the main elements of the calculation of effective range by assuming we are at UHF or lower, where the atmospheric attenuation is minimal. This is step 3. We can add the atmospheric effects later as discussed already.

\[ P_R = ERP - L_P + G_R \]

where \( P_R \) is the received power, \( ERP \) is the effective radiated power of the transmitter, \( L_P \) is the propagation loss (by whatever propagation model applies), and \( G_R \) is the receiving antenna gain.

We set the received power equal to the sensitivity, which occurs at the maximum effective range. Then we solved for the propagation loss and solve for the range element of that loss. Finally, we determine the range at which the received power equals the sensitivity.

\[ L_P = ERP + G_R - P_R = ERP + G_R - S \]

To get to the range part of the propagation loss, we need to specify the propagation model. We will consider only free-space and two-ray propagation at this time. The knife edge diffraction model is nonlinear, so we will consider that later.

The heart of the process is step 3, which starts with the propagation loss (\( L_P \)), which is either free-space loss expression from Section 5.1.1 or the two-ray loss from Section 5.1.2:

For free-space loss:

\[ L_P = L_{FS} = 32 + 20 \log(F) + 20 \log(d) \]

where \( L_{FS} \) is the free-space loss in dB, \( F \) is the operating frequency in MHz, and \( d \) is the transmitter to receiver distance in km.

Plugging this into the \( L_P \) equation:

\[ 32 + 20 \log(F) + 20 \log(d) = ERP + G_R - S \]

The range part of the propagation loss is

\[ 20 \log(d) = ERP + G_R - S - 32 - 20 \log(F) \]

The effective range is then

\[ d = \text{Antilog}\{20 \log(d)/20\} \]
For two-ray loss

\[ L_P = L_{2\text{ray}} = 120 + 40 \log(d) - 20 \log(h_T) - 20 \log(h_R) \]

where \( d \) is the link distance in km, \( h_T \) is the transmitting antenna height in meters, and \( h_R \) is the receiving antenna height in meters.

Plugging this into the \( L_P \) expression with \( P_R \) set equal to \( S \):

\[ L_P = ERP + G_R - S \]

\[ 120 + 40 \log(d) - 20 \log(h_T) - 20 \log(h_R) = ERP + G_R - S \]

So:

\[ 40 \log(d) = -120 + 20 \log(h_T) + 20 \log(h_R) + ERP + G_R - S \]

from which the effective range can be determined as

\[ d = \text{Antilog} \left\{ \frac{40 \log(d)}{40} \right\} \]

Remember (from Section 2.1) that what we call Antilog is actually raising 10 to the power of the expression in \( \{ \} \) brackets.

Now, the trick is to determine the allowable propagation loss and plug it into that formula. The allowable loss is the difference between the ERP and the signal strength arriving at the antenna (\( P_A \)) minus the atmospheric attenuation. But for the receiver to properly reproduce the information signal, the input signal to the receiver (\( P_R \)) must at least equal the receiver’s sensitivity level (\( S \)). If we require a margin (\( M \)), \( P_R \) must be above \( S \) (dBm) by \( M \) (dB). Therefore, the received power must reach the receiver with a signal strength of at least \( S + M \) (dBm). Since the receiving antenna adds \( G_R \) dB of gain, \( P_A \) must be at least \( S + M - G_R \) (dB). So the maximum acceptable propagation loss is defined by the following expression:

\[ L_P = ERP - L_{Atm} - (S + M - G_R) \]

Where:

- \( L_P \) = Propagation loss (dB)
- \( ERP \) = Effective radiated power (dBm)
- \( L_{Atm} \) = Atmospheric loss (dB)
- \( S \) = Receiver system sensitivity (dBm)
- \( M \) = Link margin (dB)
- \( G_R \) = Receiving antenna gain (dB)
For example, if the signal frequency is 250 MHz (meaning that there is negligible atmospheric loss), the transmitter ERP is +50 dBm (in the direction toward the receiver), the receiver sensitivity is –73 dBm, the receiving antenna gain is +10 dB (in the direction toward the transmitter). The height of the transmitting antenna is 2 m, the height of the receiving antenna is 30 m, and you must have 20 dB of performance margin and the effective link range (in kilometers) would be calculated as follows:

For free-space propagation:

\[
20 \log(d) = ERP + G_R - S - M - 32 - 20 \log(F)
\]
\[
= 50 \text{ dBm} + 10 \text{ dB} + 73 \text{ dBm} - 20 \text{ dB} - 32 \text{ dB} - 20 \log(250)
\]
\[
= 50 + 10 + 73 - 20 - 32 - 48 = 33
\]

Note that the sensitivity is a negative number (so becomes positive in the expression) and that the atmospheric attenuation is negligible at 250 MHz. So the effective range of the link is calculated from 20 log\(d\) as

\[
d(\text{in km}) = \text{Antilog}\{33/20\} = 44.7 \text{ km}
\]

For two-ray propagation

For two-ray propagation to apply, it is necessary to assume that both the transmitting and receiving antennas illuminate the ground. Now, plug the numbers into the two-ray range term.

\[
40 \log(d) = -120 + 20 \log(h_T) + 20 \log(h_R) + ERP - L_{Atm} + G_R - M - S
\]
\[
= -120 + 20 \log(2) + 20 \log(30) + 50 \text{ dBm} - 0
\]
\[
+ 10 \text{ dB} - 20 \text{ dB} + 73 \text{ dBm} = -120 + 6 + 29.5 + 50 - 0
\]
\[
+ 10 - 20 + 73 = 28.5
\]

from which the distance can be determined as

\[
d(\text{in km}) = \text{Antilog}\{28.5/40\} = 5.2 \text{ km}
\]

Notice it is a lot shorter range because the two-ray attenuation is a significantly greater function of range at this frequency.

5.3.4 Required Transmitter Power or Antenna Gain

It is a straightforward process to determine the required transmitter power or the gain required of either the transmitting or receiving antenna if the other link parameters are established. If all elements are expressed in dB or dBm, the transmitter power and the transmitting antenna gain sum to the effective radiated power (ERP). The ERP is then reduced by the link loss to the signal level at the receiving antenna. The receiving antenna gain is subtracted from the receiver sensitivity to determine the minimum level that the signal strength at the receiving antenna can be for proper link operation. However, to provide link margin the signal strength at the receiver must be greater than that minimum by
M dB. In an equation this is

\[ P_T + G_T - L = S - G_R + M \]

where:

- \( P_T \) = Transmitter power (dBm)
- \( G_T \) = Transmitting antenna gain (dB)
- \( L \) = Combined link losses (dB)
- \( S \) = Receiver sensitivity (dBm)
- \( G_R \) = Receiving antenna gain (dB)
- \( M \) = Link margin (dB)

Each of the following three formulas is a reorganization of this formula to solve for one of the six values in terms of the other five. The assumption in each case is that five of the six values are set because the hardware exists, the link geometry is required, or that value has been calculated earlier.

**Required Transmitter Power:**

\[ P_T = S - G_R + M + L - G_T \]

For example, if the receiver sensitivity is –80 dBm, a 20 dB margin is required, the total link losses (spreading + atmospheric) are 110 dB, the transmitting antenna has 10 dB gain, and the receiving antenna has 3 dB gain, then adequate link performance requires the transmitter power to be

\[ -80 \text{ dBm} - 3 \text{ dB} + 20 \text{ dB} + 110 \text{ dB} - 10 \text{ dB} = +37 \text{ dBm} \]

(Which is 5 W)

**Required Transmitting Antenna Gain:**

\[ G_T = S - G_R + M + L - P_T \]

For example, if the receiver sensitivity is –80 dBm, a 20 dB margin is required, the total link losses are 110 dB, the transmitter power is 10 watts (+40 dBm), and the receiving antenna has 3 dB gain, then adequate link performance requires the transmitting antenna gain to be

\[ -80 \text{ dBm} - 3 \text{ dB} + 20 \text{ dB} + 110 \text{ dB} - 40 \text{ dBm} = 7 \text{ dB} \]

**Required Receiving Antenna Gain:**

\[ G_R = S - P_T - G_T + M + L \]

For example, if the receiver sensitivity is –80 dBm, a 20 dB margin is required, the total link losses (spreading + atmospheric) are 110 dB, the transmitter power is 1 watt (+30 dBm), and the transmitting antenna has 10 dB gain, then adequate link performance requires the receiving antenna gain to be

\[ -80 \text{ dBm} - 30 \text{ dBm} - 10 \text{ dB} + 20 \text{ dB} + 110 \text{ dB} = 10 \text{ dB} \]
5.4 Interfering Signals

Whether considering unintentional interference or deliberate jamming, the following discussion will allow you to determine the ratio of a radiated interfering signal to the desired signal as seen by the receiver.

It is important to understand only what the receiver sees (as signals entering through its input connector) matters. If the interfering transmitter is at a location different from that of the transmitter producing the desired signal, the receiving antenna pattern may cause a difference in the antenna gain for the desired and interfering signals. Also, if the interfering signal is not at the exact same RF frequency as the desired signal, its interference may be attenuated by filtering anywhere in the receiving system.

For the rest of Section 5.4, we will use the following assumptions:

- The receiver tuning is centered on the desired signal.
- The receiving antenna boresight is pointed at the desired signal’s direction of arrival.
- The interfering transmitter is located remote from the transmitter generating the desired signal so that its range and angle of arrival are different.

The physical arrangement is then as shown in Figure 5-22.

The definition of three new terms will help us deal with interfering signals:

- $I/S = \text{interference-to-signal ratio, or the power ratio of the interfering signal to the desired signal at the receiver input but only as the receiver can receive it}$
- $I_A = \text{angular isolation, or the boresight gain of the receiving antenna; the gain of the receiving antenna in the direction of the interfering transmitter}$
- $I_F = \text{frequency isolation, or the amount of attenuation that all of the filters in the receiving system apply to the interfering signal, relative to the throughput of the desired signal}$

![Figure 5-22](image-url)
$I_F$ is a little tricky because frequency isolation can have several causes as shown in Figure 5-23. If the interfering signal is within the frequency range of desired signals but not at the frequency of the specific desired signal, it is generally attenuated only by filters within the receiver. If outside the frequency range of desired signals, it may be further attenuated by a band-pass filter before the receiver. If it is outside the frequency range of the receiving antenna, the antenna will act like a filter, providing additional attenuation.

In general, it is the ratio of an interfering signal to the desired signal ($I/S$) that causes trouble, since automatic or manual gain control in a receiver can generally take care of problems associated with the absolute power of interfering signals received. The ($I/S$) ratio is calculated from:

$$I/S = ERP_I - ERP_S - L_I + L_S - I_A - I_F$$

Where:

$I/S$ = interference-to-signal ratio (dB)
$ERP_I$ = ERP of the interfering signal (dBm)
$ERP_S$ = ERP of the desired signal (dBm)
$L_I$ = link losses of the interfering signal (dB)
$L_S$ = link losses of the desired signal (dB)
$I_A$ = antenna isolation (dB)
$I_F$ = filter isolation (dB)

The link losses of the interfering and desired signals can be from any of the propagation models described in Section 5.1, and there is no reason that they need to be from the same model.

For in-band interference, the $I_F$ term is zero. If all of the signals use the same propagation model it may be practical to simplify the $I/S$ equation in a number of other interesting ways, such as canceling the 20 log frequency term in free-space attenuation or the receiving antenna height term for two-ray attenuation.

Out-of-band interference is a little more complicated. It is necessary to have some information about the ultimate rejection of filters and to look at the receiving system configuration to determine which filters will let the interfering signal through and which will attenuate it. When a receiver is installed physically near a very powerful out-of-band transmitter, a band-stop filter is sometimes added specifically to bring the interference-to-signal ratio within acceptable bounds.
Antennas are well specified as to the frequencies they accept but generally make very poor filters because they are not easily designed to reject frequencies.

The signal environment shown in Figure 5-24 will be used for interference to signal calculation examples. In these exercises, we will assume that because of antenna gains all of the signals have free-space propagation.

Our receiving antenna has gain equal to:

- 10 dB toward the desired signal transmitter
- 2 dB toward interfering signal transmitter 1
- –3 dB toward interfering signal transmitter 2

The desired signal is a mobile communications radio with

- ERP = +30 dBm (toward our receiver)
- Frequency = 225 MHz
- Range = 40 km

Signal $I_1$ is from a transmitter identical to that producing the desired signal but 100 km from us. Its transmitting antenna provides an ERP of +30 dBm toward our receiver.

Signal $I_2$ is from a high-power commercial broadcast station operating at 100 MHz only 2 km from our receiver. It has ERP = +80 dBm toward our receiver.

First, calculate the $I/S$ ratio for the interfering signal $I_1$. This is an in-band interfering signal so the $I_F$ term is zero and the equation is

$$I/S = ERP_I - ERP_S - L_I + L_S - I_A$$

- $L_I$ is the free-space loss for 100 km at 225 MHz, which is $32 + 20 \log(100) + 20 \log(225) = 119$ dB.

![Figure 5-24 Interference](image_url)
- $L_S$ is the free-space loss for 40 km at 225 MHz, which is $32 + 20 \log(100) + 20 \log(225) = 111$ dB.
- $I_A$ is the difference between the receiving antenna’s boresight gain, and its gain in the direction of the interfering signal, which is $10 \text{ dB} - 2 \text{ dB} = 8$ dB.

So $I/S = 30 \text{ dBm} - 30 \text{ dBm} - 119 \text{ dB} + 111 \text{ dB} - 8 \text{ dB} = -16 \text{ dB}$

The interfering signal is received 16 dB below the desired signal by the receiver. Depending on the modulation, it might be noticeable but will not prevent proper reception of the desired signal.

Now consider the $I/S$ ratio for the interfering signal $I_2$. This is an out-of-band interference signal, so the equation is

$$I/S = ERP_I - ERP_S - L_I + L_S - I_A - I_F$$

- $L_I$ is the free space loss for 2 km at 100 MHz, which is $32 + 20 \log(2) + 20 \log(100) = 78$ dB.
- $L_S$ is the free-space loss for 40 km at 225 MHz, which is $32 + 20 \log(100) + 20 \log(225) = 111$ dB.
- $I_A$ is the difference between the receiving antenna’s boresight gain and its gain in the direction of the interfering signal, which is $10 \text{ dB} - (-3) \text{ dB} = 13$ dB.

So $I/S = 80 \text{ dBm} - 30 \text{ dBm} - 78 \text{ dB} + 111 \text{ dB} - 13 \text{ dB} - I_F = 70 \text{ dB} - I_F$

$I_F$ is not well defined for this example, so let’s use some typical numbers. The interfering signal is well out of band, so the ultimate filter rejections will probably be about 60 dB. The type of antenna was not specified but will probably not give more than 10 dB of frequency rejection, so the total $I_F$ is 70 dB.

This strong out-of-band interference signal will be equal to our desired signal, which will probably preclude proper reception. If we can’t move away from the interfering signal (a few kilometers would make a big difference), we should look into an extra high pass- or band-stop filter to reduce the $I/S$ ratio.

### 5.5 Dynamic Range

Dynamic range is the difference (in dB) between the strongest and weakest signals a receiver or receiving system can accept without degrading its performance. With manual or automatic gain control, a receiver’s total dynamic range will allow it to accept a very wide range of signal power. This is no problem as long as the desired signal is the strongest (or nearly the strongest) signal that will be received. However, if the receiver must accept a much weaker signal in the presence of stronger in-band signals, its instantaneous dynamic range becomes important.

The instantaneous dynamic range is the difference (in dB) between the strongest and weakest signals that can be present in a receiver’s passband while...
the receiver is meeting its full specified performance in receiving and processing
the weaker signal.

Whether the total or instantaneous dynamic range definition is used, the
maximum strength (in dBm) at which a receiving system can receive and process a
signal is the system sensitivity plus the dynamic range. For example, if a receiving
system sensitivity is \(-90\) dBm and its dynamic range is 60 dB, the strongest signal it
can accept is \(-90\) dBm + 60 dB = \(-30\) dBm.

The calculation of dynamic range is covered in Chapter 6.
Chapter 6
Receiver Sensitivity and Dynamic Range

Sensitivity and dynamic range have to do with the range of received signal strength that any kind of receiving system can accept. Sensitivity describes the weakest signals that can be received and processed, and dynamic range deals with the strongest signals that can be present. This chapter details how to calculate both values for analog and for digital receivers.

6.1 Sensitivity

The sensitivity of a receiver defines the weakest signal it can receive and still provide an output of adequate quality. Sensitivity specifications are most commonly stated in dBm and are usually negative numbers because the sensitivity is almost always less than 1 milliwatt.

An important issue is where to define sensitivity. To accurately predict link performance, all of the gains and losses in the receiving part of the link must be counted. Therefore, system sensitivity is usually defined at the output of the antenna as shown in Figure 6-1.

It is the system sensitivity that has been used in the link equation formulas in Chapter 5. This is important because when the antenna gain (in dB) is subtracted from the system sensitivity (in dBm) we should get the signal strength which must arrive at the antenna to allow the link to work properly.

If for some reason you want to define the system sensitivity at some other point (e.g., at the end of a cable from the antenna) you can certainly do so, but you’ll need to account for the cable loss as a reduction in the antenna gain if you want to use the Chapter 5 formulas.

As you will see later, the first element in the system that provides gain has a dramatic effect on the sensitivity, so you should always define the system sensitivity before the first amplifier when evaluating link performance.

The sensitivity of any receiver or receiver system is determined by the sum of three numbers:

- Thermal noise level (kTB)
- Noise figure (NF)
- Required prediction signal-to-noise ratio (SNR)
When defining sensitivity in dBm, the sensitivity is related to these three numbers as shown in Figure 6-2. Since numbers expressed in dB are logarithmic, the SNR (in dB) is found by subtracting the noise level (in dBm) from the signal level (in dBm). Since the sensitivity in dBm is normally a negative number, a bigger (i.e., more negative) number equals a smaller signal. (Lots of very smart people get in trouble over that one.)

$kTB$ is the basic thermal noise level in the receiver. It is a function of the operating temperature and the bandwidth. The noise figure is the amount of noise that the receiver system adds. The required predetection SNR is a function of the output signal quality required. It is the SNR at the system sensitivity definition point (i.e., the output of the antenna) that must be used in calculating sensitivity, and as you will see in Section 6.1.3 this can be quite different from the output signal SNR for some types of modulation.

It is important to be sure that all three of these components of sensitivity are referenced to the same point in the system: the point at which you are defining the sensitivity.

### 6.1.1 Thermal Noise Level

Thermal noise level ($kTB$) defines the thermal noise present in an ideal receiver—the lowest noise that can be theoretically achieved. It is a function of the effective
receiver bandwidth and the operating temperature. \((k\) is Boltzmann’s constant, \(T\) is the system operating temperature, and \(B\) is the system bandwidth.)

A commonly used number for \(kT\) is –114 dBm per MHz, which is the thermal noise level for a receiver with a 1 MHz bandwidth operating at standard temperature (defined as 290 K, or about 63° F or 17° C). Appendix A includes the derivation of this number in case you get a serious case of urge for rigorousness.

\(kT\) is not very sensitive to temperature changes. The operating temperature has to rise to 92° C (i.e., 198° F, almost the boiling point of water) to increase \(kT\) by 1 dB. However, it is very sensitive to receiver bandwidth and varies by 90 dB over the commonly used range of bandwidths. You can calculate \(kT\) at standard temperature using

\[
kTB = -114 \text{ dBm} + 10 \log(\text{Receiver Bandwidth}/1 \text{ MHz})
\]

or you could just read it from the chart in Figure 6-3.

To adjust \(kT\) for operating temperature, use

\[
kTB = \text{Standard temperature } kTB + 10 \log(\text{Temperature in degrees K}/290)
\]

### 6.1.2 Noise Figure

The second element of sensitivity is noise figure (NF). It defines the amount of noise the receiver or receiving system adds to received signals (above \(kT\)). System noise figure is a number (typically between 3 and 18 dB) derived from the noise figures of the system’s amplifiers and receivers and system losses.
The noise figure of an amplifier or a receiver is specified by the manufacturer (in dB). It is determined by measuring the noise level output by the device but is referenced to the input of the device. Thus, the noise figure is the amount of additional noise that would be input to a perfect, noiseless device to produce the measured level of output noise.

An amplifier’s noise figure is its output noise in excess of $kT\beta$ reduced by the gain. A receiver’s noise figure includes the effects of all receiver processing but is still the amount of noise in excess of $kT\beta$ which would have to be added at the receiver input to cause its output noise level if the receiver were noiseless.

The noise figure, which is summed with $kT\beta$ and SNR in the sensitivity calculation, is the system noise figure. This is the amount of noise that every element of the system adds but all referenced to the point at which sensitivity is defined (typically at the antenna output).

The system noise figure is very much a function of the system configuration. The simplest case is shown in Figure 6-4. Here we have a receiver attached to an antenna by a cable and perhaps passing though some switches, power dividers, or other passive devices. Passive means that they have no gain or other nonlinear throughput characteristics. (Switches are OK because they are linear when they are not switching). In this case, the system noise figure is the sum of the receiver noise figure (in dB) and all losses (in dB) between the antenna and the receiver input.

$$\text{System NF} = \text{Passive Losses} + \text{Receiver NF}$$

For example, if a receiver with 8 dB noise figure is attached to an antenna by a long cable that has 5 dB of loss, the system noise figure is 13 dB.

For more complex systems involving a preamplifier, the system noise figure can be determined from the graph in Figure 6-5. To use this graph, add the preamplifier gain and noise figure, and then subtract the losses before the receiver. Run a horizontal line through this value on the left side of the graph. Then run a vertical line through the receiver noise figure at the bottom of the graph. The intersection of these two lines defines the system noise figure degradation from everything down stream of the input to the preamplifier. The system noise figure is then calculated as

$$\text{System NF} = \text{Loss before Preamp} + \text{Preamp NF} + \text{Degradation}$$

Figure 6-6 shows some typical values for the block diagram at the top of Figure 6-5. The preamplifier noise figure is 5 dB, the preamplifier gain is 20 dB, and the loss before the receiver is 8 dB ($20 + 5 - 8 = 17$ dB). The receiver noise
Preamplifiers are used primarily because loss before the preamplifier directly reduces the sensitivity whereas loss after the preamplifier degrades sensitivity only by a (usually much smaller) degradation factor.
For more complex systems, with more than one preamplifier, start at the receiver and work back toward the front end. First determine the system noise figure at the input of the last preamplifier using the aforementioned procedure. Then repeat the procedure using the system noise figure at this point as the receiver noise figure while determining the system noise figure looking into the next earlier preamplifier.

6.1.3 Required SNR

The hardest part of defining the sensitivity, at least the part over which the loudest arguments take place, is the definition of adequate quality. The problem is that adequacy depends on the use to which the information passed by the link is to be put. To be useful in sensitivity definition, adequacy must first be reduced to an SNR for the information signal output by the receiver. Then it must be converted to an SNR at the antenna output (which is the sensitivity definition point). To differentiate these two SNR values (which can be quite different) we will call the signal-to-noise ratio at the sensitivity definition point the RF SNR.

Table 6-1 shows typical required output SNR values for various types of signals and the equivalent RF SNR for the modulations indicated. This table gives only typical values. The actual output SNR required will depend on the specific application. For example, laboratory studies have shown that 8 dB is (barely) adequate video SNR for trained operators to quickly determine which of several shapes appear on the screen; selection accuracy does not improve with more SNR, but the picture quality improves up to 16 dB. Note that at 8 dB the picture has so much snow that you would probably do violence to a commercial television set with that signal quality.

<table>
<thead>
<tr>
<th>Type of Signal</th>
<th>Output SNR</th>
<th>RF SNR</th>
</tr>
</thead>
<tbody>
<tr>
<td>Amplitude-Modulated Signals</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Voice Communications</td>
<td>12 dB</td>
<td>12 dB</td>
</tr>
<tr>
<td>High-Quality AM Radio</td>
<td>30 dB</td>
<td>30 dB</td>
</tr>
<tr>
<td>Analog TV Broadcast</td>
<td>40 dB</td>
<td>40 dB</td>
</tr>
<tr>
<td>Pulse Signal (Manual Processing)</td>
<td>8 dB</td>
<td>8 dB</td>
</tr>
<tr>
<td>Pulse Signal (Automatic Processing)</td>
<td>15 dB</td>
<td>15 dB</td>
</tr>
<tr>
<td>Frequency-Modulated Signals</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Voice Communications</td>
<td>12 dB</td>
<td>12 dB</td>
</tr>
<tr>
<td>High-Quality FM Radio</td>
<td>30 dB</td>
<td>12 dB</td>
</tr>
<tr>
<td>High-Quality TV Broadcast</td>
<td>40 dB</td>
<td>12 dB</td>
</tr>
<tr>
<td>Receiver Using PLL Discriminator</td>
<td>20 dB</td>
<td>4 dB</td>
</tr>
<tr>
<td>Digitally Encoded Signals</td>
<td></td>
<td></td>
</tr>
<tr>
<td>(Including Digital TV)</td>
<td>Depends on 12 dB</td>
<td>quantizing</td>
</tr>
</tbody>
</table>

*Table 6-1 The typical required output SNR values for various types of signals and equivalent RF SNR for the modulations indicated*
The following sections will allow you to calculate the RF SNR from the output SNR (and vice versa) for your specific application with the applicable modulation.

**Required SNR for AM and Narrowband FM Signals**

Neither amplitude modulation nor narrowband frequency modulation provides any improvement in output SNR relative to the RF SNR. Therefore, the two SNR figures are identical for these modulations.

**Required SNR for FM Signals**

Frequency modulation gives performance advantages by spreading the transmitted signal over a frequency band that is wider than the bandwidth of the information signal. The ratio of the transmission bandwidth to the information bandwidth is called the modulation index (\(\beta\)). The SNR of the information signal output from the receiver can be greater than the RF SNR by a factor that is a function of \(\beta\). However, to get this improvement factor the RF SNR must be above a threshold factor that depends on the type of FM discriminator used. For the most common type of discriminator, the threshold is about 12 dB, while a phase locked loop (PLL) discriminator will provide the improvement above an RF SNR threshold level of about 4 dB. Providing that the threshold level is met, the signal-to-noise improvement factor \((IF_{FM})\) is given by

\[ IF_{FM} = 5 + 20 \log(\beta) \]

(For RF SNR > Threshold)

For example, if the modulation index is 5, the improvement factor is

\[ IF_{FM} = 5 + 20 \log(5) = 5 + 14 = 19 \text{ dB} \]

So if the receiver has a normal FM discriminator, a signal with a modulation index of 5 that is received with a 12 dB RF SNR will produce an output signal with SNR = 31 dB.

**Required SNR for Digital Signals**

Digitized signals are different from analog information signals in that the output SNR is a function of the way the signal was digitized. The noise is not really noise but is instead distortion caused by the quantizing steps in the digitizing process. This quantizing noise is so called because it looks and sounds much like noise to operators. RF noise causes the receiver to generate bad bits, or bit errors. These do distort the output information signal, but digital systems are usually specified directly in terms of the bit error rate since many such systems carry information that has never been in analog form.

### 6.2 Digitized Analog Signals

Figure 6-7 shows a digitized analog signal and what it looks like when it is reconstructed. This figure and the following discussion assume that pulse code
modulation (PCM) is used. This is the most common of several digitization techniques that can provide application-specific advantages, but all give approximately the same throughput quality for the same bit rate.

The two primary considerations in PCM digitization are the number of quantizing levels and the sampling rate.

6.2.1 Sampling Rate

The sampling rate must be at least twice the highest information signal frequency to capture all of the signal’s information. For a simple sine wave, two samples per cycle are enough. A low-pass filter will smooth those two samples per cycle in the output to a perfect sine wave. However, if the curve varies from sinusoidal shape (as the curve in Figure 6-7 definitely does) there are higher frequency components present. The point is that the sampling rate must be high enough to capture any character of the curve that you want to reproduce.

6.2.2 Quantizing Levels

Assuming that there is an adequate sampling rate, the number of digitizing bits per sample determines the signal-to-quantizing ratio (SQR). The number of quantizing levels is \(2^n\), where \(n\) is the number of bits per sample, and the SQR is given by the following formula, which rounds to 1 dB:

\[
SQR \text{ (in dB)} = 5 + 3(2n - 1)
\]

where \(n\) = the number of bits per sample.
For example, with six bits per sample, the SQR is

\[ SQR = 5 + 3(11) = 38 \text{ dB} \]

Table 6-2 shows the number of quantizing levels and the SQR (in dB) for each number of bits per sample up to the limit of common analog-to-digital converters (ADCs).

6.2.3 RF SNR

The effect of the RF SNR is that the noise causes bit errors. The modulator in the transmitter has converted the digital signal (which is a series of ones and zeros) into some sort of modulation medium as described in Section 4.2.2. A demodulator in the receiver will convert the received modulation back into a digital bit stream, but the addition of noise to the signal adds uncertainty as shown in Figure 6-8. This figure shows a normally distributed probability density for the modulation value that the signal plus noise could have. For example, if the digital information were carried on a frequency-shift key (FSK) modulation, the abscissa would be frequency, the dark arrows represent the one and zero signal frequencies, and the bell-shaped curves the probability that a one or zero signal (along with noise) would be at each frequency. The error zones (i.e., a one detected as a zero and a zero detected as a one) are shaded.

Although the bit error rate versus SNR depends on the specific RF modulation used, Figure 6-9 shows the bit error rate versus RF SNR for two types of modulation that span the values of most common modulations. Note that this chart assumes that the ratio of the effective bandwidth of the receiver to the signal bit rate is unity. One point of interest is that on the lower part of the curve (where most digital systems are designed to operate) the slope of the lines is about 1 dB per order of magnitude of bit error rate. To pick an example from the chart, a non-coherent FSK modulated digital signal with 11 dB RF SNR would achieve a little less than $10^{-3}$ bit error rate.

The bit error rate is sometimes plotted against Eb/No. (Eb is the product of signal power and bit period.) Eb/No is not actually a signal-to-noise ratio, but it determines the RF SNR that would be achieved if the signal were passed through an optimum matched filter for the modulation chosen.
By definition, Eb/No is the RF SNR multiplied by the ratio of the receiver effective bandwidth to the signal bit rate.

\[
\text{Eb/No} = \text{RF SNR} \times \left( \frac{\text{receiver effective bandwidth}}{\text{signal bit rate}} \right)
\]

If the bandwidth to bit rate ratio is unity, the abscissa of Figure 6-9 graph can be written as the RF SNR (in dB). This is done in some communication theory textbooks.

### 6.3 Dynamic Range

Dynamic range is the difference (in dB) between the strongest and weakest signals a receiver or receiving system can accept without degrading its performance. With manual or automatic gain control, a receiver’s total dynamic range will allow it to accept a very wide range of signal power. This is no problem as long as the desired signal is the strongest (or nearly the strongest) signal that will be received. However, if the receiver must accept a much weaker signal in the presence of stronger in-band signals, its instantaneous dynamic range becomes important.

The instantaneous dynamic range is the difference (in dB) between the strongest and weakest signals that can be present in a receiver’s pass band while the receiver is meeting its full specified performance in receiving and processing the weaker signal.

Whether the total or instantaneous dynamic range definition is used, the maximum strength (in dBm) at which a receiving system can receive and process a signal is the system sensitivity plus the dynamic range. For example, if a receiving system sensitivity is −90 dBm and its dynamic range is 60 dB, the strongest signal it can accept is

\[
-90 \text{ dBm} + 60 \text{ dB} = -30 \text{ dBm}
\]
6.3.1 Analog versus Digital Dynamic Range

Since many digital receivers are in use, it is important to remember that digital receivers comprise an analog front end followed by an ADC and then a computer in which the digital receiver functions are performed (Figure 6-10). The analog and digital sections each have defined dynamic ranges. First, consider the dynamic

![Dynamic range of RF and digital parts of receiver must be compatible](image)

**Figure 6-10** Dynamic range of RF and digital parts of receiver must be compatible

**Figure 6-11** Intercept point chart

![Intercept point chart](image)
range of the analog section and then that of the digital section. The two receiver sections should have the same dynamic range.

### 6.3.2 Analog Receiver Dynamic Range

The dynamic range is typically determined by a preamplifier, which is specified in terms of its gain, noise figure, and intercept points. The impact of preamplifier gain and noise figure on system sensitivity was discussed earlier. The intercept points of the preamplifier impact the system dynamic range.

Figure 6-11 is used to determine the dynamic range from the intercept points. This diagram relates to the output of the preamplifier. The ordinate and abscissa of the graph are both logarithmic scales of preamplifier output power in dBm. The fundamental line represents the output power of a single amplified signal from the amplifier. Note that it has a 1:1 slope. The second-order response line represents the second-order spurious response at the output of the amplifier. This is the level of the spurious signal produced at the output of the amplifier at a frequency double the fundamental frequency or at the sum or difference frequencies of two input signals. The second-order response line has a 2:1 slope and intersects the fundamental line at what is called the second-order intercept point (IP2). Starting at the

![Figure 6-12 Second and third order intercept points](image)

*Figure 6-12 Second and third order intercept points*
output power of the one or two signals causing the spurious response on the abscissa of the graph, move up to the second-order line and then left to the ordinate to read the level of the second-order spurious response output.

The third-order response line is the level of third-order spurs. A third-order spurious response is output at two times the frequency of one input signal plus or minus the frequency of the second input signal (or one frequency plus or minus twice the other frequency). This line has a 3:1 slope and intersects the fundamental line at the third-order intercept point (IP3).

The intercept points are the signal levels at which the spurious response lines intercept the fundamental line in Figure 6-12. For the example shown, the second-order intercept point is +50 dBm, and the third-order intercept is at +20 dBm. Two input signals at –27 dBm would produce third- and second-order spurious outputs at –100 dBm and –112 dBm, respectively, as shown in Figure 6-13.

A receiver design will normally eliminate the second-order spurs by selection of intermediate frequencies and multiple conversions if necessary. However, the third-order spurs often cannot be avoided and limit the receiver’s spurious free dynamic range.

The graph shows an intercept validity limit line. This is the level at which the amplifier is no longer well behaved. It is near the point at which compression in the

![Figure 6-13 Second and third order spur isolation](image-url)
amplifier output (caused by saturation) causes the actual levels of the spurious outputs to vary significantly from the second- and third-order line values. Thus, the dynamic range must be calculated to the left of this validity limit.

6.3.3 Determining Dynamic Range

Figure 6-14 adds some information to the graph in Figure 6-13. The sensitivity line is the level of a preamplifier output caused by a received signal (from the antenna) at the receiver system sensitivity level. (At the preamplifier output, the sensitivity signal level is the level at the antenna output adjusted for all gains and losses between the antenna and the preamplifier output.) In this example, the sensitivity level preamplifier output signal is –100 dBm. To calculate the dynamic range limited by third-order spurs, draw a vertical line from the intersection of the third-order intercept line and the sensitivity line up to the fundamental line. The vertical distance between these two lines (i.e., the difference in output signal strength in dB) defines the output signal strength of strong signals that will cause spurs at the sensitivity level. This means that if the signals causing spurs are at the fundamental line level (–32 dBm), the third-order spurious outputs will be at –100 dBm.

Figure 6-14  Spur isolation lines intersecting sensitivity level
(which is the sensitivity level of the receiver at this point). So the dynamic range as shown in Figure 6-15 is 68 dB.

6.4 Digital Dynamic Range

The dynamic range of the digital part of the receiver depends on the number of bits produced by the ADC. The weakest signal is digitized as one in the least significant bit (all other bits being zeros), while the strongest signal is digitized as all ones. Figure 6-16 shows the digitization of the maximum and minimum measurable levels in a 4-bit ADC.

The dynamic range (in dB) is then

$$DR = 20 \log_{10}(2^n)$$

where $n$ is the number of bits to which the input signal is digitized.

Note that the conversion to dB has a 20 multiplier rather than the 10 multiplier when signal power ratios are converted. This is because the digitizer quantizing levels that determine the digital word produced are voltages. For example 10 bits provides 60 dB dynamic range. Table 6-2 shows the digital dynamic range as a function of the number of digitizing bits.
Maximum measurable amplitude

Minimum measurable amplitude

Figure 6-16  Digital receiver dynamic range

Table 6-2  Dynamic Range versus Digitizing Bits

<table>
<thead>
<tr>
<th>Number of Bits</th>
<th>Dynamic Range (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>4</td>
<td>24</td>
</tr>
<tr>
<td>5</td>
<td>30</td>
</tr>
<tr>
<td>6</td>
<td>36</td>
</tr>
<tr>
<td>7</td>
<td>42</td>
</tr>
<tr>
<td>8</td>
<td>48</td>
</tr>
<tr>
<td>9</td>
<td>54</td>
</tr>
<tr>
<td>10</td>
<td>60</td>
</tr>
<tr>
<td>11</td>
<td>66</td>
</tr>
<tr>
<td>12</td>
<td>72</td>
</tr>
<tr>
<td>13</td>
<td>78</td>
</tr>
<tr>
<td>14</td>
<td>84</td>
</tr>
<tr>
<td>15</td>
<td>90</td>
</tr>
<tr>
<td>16</td>
<td>96</td>
</tr>
</tbody>
</table>
Chapter 7
Challenging Conditions

This chapter covers some of the challenging conditions we deliberately ignored in the first six chapters:

- Attenuation for transmission through rain or fog
- Non–line of sight transmission
- The Doppler effect
- Antenna misalignment

7.1 Rain and Fog Attenuation

Radio signals are attenuated as they pass through rain and fog—in excess of the normally accepted atmospheric attenuation levels described in Chapter 5. The bad news is that the charts for rain attenuation given in different references disagree with each other by a dB or so. The good news is that for practical purposes the absolute numbers are not nearly as important as an understanding of what is going on and the relative rain losses among various alternative link designs.

Anyone who has ever been out in the rain knows that the density of rainfall in any one location changes from minute to minute and that a few yards away may be much greater or less. Therefore, we need to establish a reasonable rainfall model and use it to determine the rain attenuation for each of multiple link configurations if we are to compare their performance in bad weather. It is also good practice to use the same model to determine how much link margin to allow to assure dependable link operation in bad weather.

Figure 7-1 shows the amount of additional attenuation per unit distance (km) caused by various densities of rainfall and fog. This chart is representative of several found in various references. Appendix E gives equivalent charts for attenuation per statute mile and per nautical mile for your convenience.

To use the chart in Figure 7-1, enter it from the bottom at the link operating frequency, go up to the line corresponding to the rainfall or fog density, and then go left to the attenuation per kilometer. The additional attenuation is this number multiplied by the path length (km) over which the link is subject to that rain or fog density.
As an example, consider a 15 GHz link passing through weather conditions as shown in Figure 7-2. Notice that the total transmission path length is 50 km. It passes through 10 km of heavy rain and 40 km of light rain.

From Figure 7-1, you can see that the rain attenuation at 15 GHz is 0.035 dB per kilometer for light rain and 0.75 dB per kilometer for heavy rain, so the total rain attenuation for the transmission path in Figure 7-2 is

- 40 km \times 0.035 \text{ dB/km} = 1.3 \text{ dB}
- 10 \text{ km} \times 0.73 \text{ dB/km} = 7.3 \text{ dB}
- Total attenuation = 8.6 \text{ dB}

(The scales in Figure 7-1 are logarithmic, so halfway between two tics is a third of the numeric value.)

Line E in Figure 7-1 shows that 10 km through a typhoon would cause 90 dB of attenuation at 15 GHz.
7.2 Non–Line of Sight Transmission

Up to now, we have assumed that line of sight conditions existed—that is, that the transmitting and receiving antennas can see each other, or nearly so for the case of knife edge diffraction (KED). In fact, transmission can take place without direct line of sight in several different ways depending on the frequency of transmission and the operating range.

At very low, low, and medium frequencies (below 3 MHz) extremely long-range propagation by ground wave is common. At high frequencies transmissions are reflected from the ionosphere, causing hops that can propagate clear around the earth given the proper conditions.

Tropospheric scatter propagation is also used for beyond the horizon transmission of very high-power signals, particularly in military applications.

Because the propagation by all of these mechanisms varies greatly with the specifics of transmitter and receiver site locations, time of day, or the height of ionospheric layers, the reader is referred to the many handbooks that cover these in adequate detail. No attempt is made here to provide simplified formulas because they just don’t work.

That leaves KED and diffraction over a spherical surface, which apply in a wide range of practical applications and for which reasonably accurate nomographs exist. The author has used them in field tests and found the measured results to be reasonably consistent with their predictions.

7.2.1 Knife Edge Diffraction

KED is one of the three models discussed in Chapter 5, so we won’t repeat the basics here. However, we will talk about the effective range and the effects of interfering signals when this model applies to one or more of the links.

It is important to remember that the transmitter must be closer to the knife edge (or to the real-life mountain ridge) than the receiver. Otherwise, the losses are very high because only tropospheric scattering applies and causes many tens of dB of loss.
Because the KED geometry eliminates the ground reflection path in the two-ray propagation model, the KED link adds the KED loss to free-space attenuation. At VHF or low UHF frequencies and relatively short ranges, free-space attenuation is significantly less than two-ray attenuation. Also, the KED loss is fairly low in that frequency range. This means that a KED link can have significantly less attenuation than a two-ray link under those conditions. This creates the surprising result that the received signal power can go up when a transmitter in a two-ray link is moved behind a ridge line—as long as the receiver is at least as far from the ridge line as the transmitter. This is counterintuitive, so it can scare you until you understand the math.

It is difficult to directly calculate the effective range of a KED link because any change in the position of the transmitter or receiver will change the KED attenuation. Therefore, it is normally necessary to reiteratively calculate the free-space loss + the KED loss for a series of link geometries, incrementally moving the transmitter or the receiver until the received power equals the receiver sensitivity.

7.2.2 Diffraction Loss over a Smooth Sphere

Figure 7-3 shows the attenuation (above line of sight spreading loss) that can be expected for transmission around a smooth spherical earth from low antennas. For antennas many wavelengths above the surface of the sphere, this will give about 2 dB too much loss.

In the example shown in the figure, a signal at 500 MHz transmitted 40 km over smooth earth would have about 12 dB of additional attenuation.

![Figure 7-3 Diffraction loss over smooth earth](image)
7.3 Doppler Effect

When there is relative motion between a transmitter and a receiver, the received frequency is different from the transmitted frequency by a factor proportional to the relative velocity. Figure 7-4 shows a transmitter directly approaching a receiver and the formula for the resulting change in frequency. Note that the received frequency is greater than the transmitted frequency because the transmitter is moving toward the receiver. Also, only the rate of change of distance between the transmitter and receiver matters. The same formula would apply if it were the receiver moving.

Figure 7-5 shows the more general case in which both the transmitter and receiver are moving in arbitrary directions. In this case, the relative velocity of the transmitter toward the receiver is the component of its velocity along the line connecting the transmitter and receiver (which is equal to its velocity times the cosine of the angle between its velocity vector and the receiver). Likewise, the receiver’s velocity must be reduced by the cosine of the angle between its velocity vector and the receiver.

\[
\Delta F = F_R - F_T = \frac{V}{C} F_T
\]

Where:
- \( F_T \) = The transmitted frequency
- \( F_R \) = The received frequency
- \( V \) = The relative velocity (toward each other)
- \( C \) = The speed of light

**Figure 7-4  Doppler effect**

\[
\Delta F = \frac{F_T}{C} (V_T \cos(\theta_T) + V_R \cos(\theta_R))
\]

**Figure 7-5  Doppler effect for arbitrary velocity vector**
vector and the transmitter to find the component that contributes to the Doppler effect. This is the true spherical angle.

Figure 7-6 shows a right spherical triangle with the three sides representing the horizontal and vertical angular offset angles between the velocity vector of a moving transmitter and receiver and the signal transmission vector to the other end of the link. The true spherical offset angle that applies to the Doppler frequency shift formula is found from

\[
\cos(\text{true offset angle}) = \cos(\text{vertical offset angle}) \times \cos(\text{horizontal offset angle})
\]

as shown in Figure 7-6.

If the transmitter and receiver are directly approaching each other, the value of each cosine becomes one (1), so the relative velocity is the sum of the two velocities. This collapses the formula in Figure 7-5 and the spherical trigonometric formula associated with Figure 7-6 to the simpler formula in Figure 7-4.

If the transmitter and receiver are located together (as in a radar) and the signal reflects from an object with relative motion, the Doppler effect applies both to the outgoing and returning transmission paths. This is called the two-way Doppler effect, and the difference frequency is proportional to twice the relative velocity. For the two-way Doppler case, just multiply the right side of each equation by two.

### 7.4 Antenna Misalignment

If a parabolic antenna is pointed directly at the antenna at the other end of the communication link, the transmitting or receiving gain of that antenna is the boresight gain. If it is misaligned, the gain will be reduced. If the misalignment is greater than the boresight to first null angle, the antenna at the other end of the link is in a side lobe, so the average side lobe gain of the antenna is used. However, if the antenna is misaligned by a small amount (and is within the main beam),
the convenient formulas can be used to determine the amount of gain reduction (see Figure 7-7).

The following formula gives the 3 dB beamwidth as a function of wavelength and antenna diameter:

$$\alpha = \frac{70\lambda}{D}$$

where $\alpha$ is the 3 dB beamwidth in degrees, $\lambda$ is the wavelength in meters, and $D$ is the diameter of the antenna in meters.

If it is more convenient to input operating frequency than wavelength, the formula becomes

$$\alpha = \frac{21000}{(D \times F)}$$

where $F$ is the operating frequency in MHz.

The formula for the gain reduction as a function of the error angle and the 3 dB beamwidth (for relatively small offset angles) is

$$\Delta G = 12\left(\frac{\theta}{\alpha}\right)^2$$

where $\Delta G$ is the gain reduction in dB because of antenna misalignment. $\theta$ is the antenna pointing accuracy in degrees and $\alpha$ is the 3 dB beamwidth.

A convenient dB formula for the gain reduction as a function of frequency, antenna diameter, and antenna pointing accuracy is

$$\Delta G = -0.565 + 20 \log(F) + 20 \log(D) + \theta^2$$

where $F$ is the operating frequency in MHz, and $D$ is the antenna diameter in meters.
Appendix A
Derivation of Equations and Charts

The following derivations are included both to show the pedigree of some of the simplified equations in this book, and more importantly, to clearly identify the assumptions and rounding that have been applied.

\[ kTB = -114 \text{ dBm/MHz} \]

- \( k = \) Boltzman’s constant = \( 1.38 \times 10^{-23} \) Joule/oK
- \( T = 290^\circ \text{K} \)
- \( B = 1 \times 10^6 \) Hz (units of Hz are 1/sec)

\[ kTB = (1.38 \times 10^{-23})(290)(10^6) \frac{\text{Joule}}{^\circ \text{K}} \frac{^\circ \text{K}}{\text{sec}} \]
\[ = 4.002 \times 10^{-15} \text{ Watts} \]
\[ = (4.002 \times 10^{-15})(10^3) \frac{\text{Watts}}{\text{milliwatts}} \]
\[ = 4.002 \times 10^{-12} \text{ mW} \]

\[ 10 \log(4.002 \times 10^{-12}) = -113.9772 \text{ dBm} \approx -114 \text{ dBm} \]

**Conversion of formulas from metric to other units**

- 3.28 ft/m  \( 10 \log(3.28) = 5.159 \text{ dB} \approx 5.2 \text{ dB} \)
- 1.6 km/mi  \( 10 \log(1.6) = 2.04 \text{ dB} \approx 2 \text{ dB} \)
- 1.15 mi/nmi  \( 10 \log(1.15) = .607 \text{ dB} \approx .6 \text{ dB} \)

**Free Space Loss = 32 + 20 \log(F) + 20 \log(d)**

For frequency in MHz and distance in km

This formula is the ratio of the surface of a sphere of radius \( d \) (the distance from the isotropic transmitting antenna) and the effective area of the isotropic receiving antenna. The surface area of the sphere is \( 4\pi d^2 \), and the receiving antenna area is \( \lambda^2/4\pi \). The ratio of these two areas is then:

\[ L_S = \frac{(4\pi)^2d^2}{\lambda^2} \]
Plugging in $\lambda = c/F$ gives:

$$\frac{(4\pi)^2d^2F^2}{c^2} \frac{m^2}{m^2} \frac{\text{sec}^2}{\text{sec}^2} = 1.755 \times 10^{-15}$$

which is unitless as a well-behaved ratio should be, but to input frequency in MHz and distance in km, we need to put in the conversion factors $10^3 \text{m/km}$ and $10^6 \text{Hz/MHz}$. Both factors are squared and in the numerator, yielding:

$$\text{LS} = 1.755 \times 10^3 (# \text{ of MHz})^2 (# \text{ of km})^2$$

This converts to dB as: $32.44 + 20 \log(F) + 20 \log(d)$ which rounds to:

$$\text{LS} = 32 + 20 \log(F) + 20 \log(d) \text{ for MHz & km}$$

Using the unit conversion formulas on page 117, the constant becomes $32.44 + (2.04 \times 2)$ (because of $d^2) = 36.52$ for distance in statute miles, so:

$$\text{LS} = 37 + 20 \log(F) + 20 \log(d) \text{ for MHz & sm}$$

and for distance in nautical miles, the constant becomes $36.52 + (.607 \times 2) = 37.734$ so:

$$\text{LS} = 38 + 20 \log(F) + 20 \log(d) \text{ for MHz & nm}$$

2 Ray Loss in dB = $120 + 40 \log(d) - 20 \log(h_T) - 20 \log(h_R)$

For distance in km and antenna heights in meters.

This formula is based on the partial cancellation of the direct ray from the transmitting antenna to the receiving antenna by the (delayed) second ray which bounces off of the ground.

The standard textbook form of this equation is:

$$L_{2\text{RAY}} = \frac{d^4}{h_T^2 h_R^2}$$

The distance and antenna height values are all in meters, so the loss is a pure unitless ratio.

Changing the distance units to km requires adding the conversion factor of $1,000$ (meters per kilometer) raised to the fourth power (i.e., $10^{12}$ which is 120 dB) to the numerator.

Converting the equation to dB form gives:

$$L_{2\text{RAY}} \text{ in dB} = 120 + 40 \log(d) - 20 \log(h_T) - 20 \log(h_R)$$

$FZ = (h_T \times h_R \times F)/24,000$

For Fresnel Zone (FZ), distance in km, antenna heights in meters, and frequency in MHz.
The distance and antenna height values are all in meters, so the loss is a pure unitless ratio.

The standard textbook form of this equation is

$$FZ = \frac{4\pi h_T h_R}{\lambda}$$

with FZ, the antenna heights, and the wavelength all in meters.

Plugging in $\lambda = c/F$ gives:

$$FZ = \frac{4\pi h_T h_R F}{c}$$

with F in Hz and $c = 3 \times 10^8$ m/sec.

If we input the frequency in MHz and read the FZ distance in km, a factor of $10^6/10^3$ is added to the right side numerator. The speed of light is $3 \times 10^8$ m/sec.

Combining the numerical terms on the right side of the equation:

$$\left(10^3 4\pi\right)/3 \times 10^8 = 4.188 \times 10^{-5}$$

Inverting this factor, so it can go in the denominator gives 23,878 which rounds to 24,000.

Now the FZ equation can be written:

$$FZ = (h_T \times h_R \times F)/24,000$$

with FZ in km, the antenna heights in meters and the frequency in MHz.

**SQR in dB** = $5 + 3(2m - 1)$

where an analog signal is digitized with m bits.

The standard textbook form of this equation is:

$$\frac{S}{N} = 3 \times 2^{(2m-1)}$$

Converting this to dB form gives:

$$10 \log(3) + (2m - 1) \times 10 \log(2)$$

$$= 4.77 + 3(2m - 1) \approx 5 + 3(2m - 1) \quad \text{dB}$$

**IFFM in dB** = $5 + 20 \log(\beta)$

For FM signals with RF SNR above threshold
The standard textbook form of this equation (assuming proper filtering) is:

\[ S_{O}/N_{O} = 3 \beta^2(S_{RF}/N_{RF}) \]

The FM improvement factor is the ratio of the output and RF signal-to-noise ratios (SNR), or \(3 \beta^2\).

Converting this to dB form gives: \(10 \log(3) + 20 \log(\beta)\)

\[ = 4.77 + 20 \log(\beta) \approx 5 + 20 \log(\beta) \]

**Antenna area = 38.6 + G − 20 \log(F) \quad \text{dBsm}**

\[ A = \frac{G\lambda^2}{4\pi} \]

Substituting \(\lambda = c/F\) and \(10^6 \text{ Hz/MHz}\) gives:

\[ A = \frac{c^2 G}{4\pi F^2} \frac{m^2 \text{ sec}^2}{(10^6 \text{ Hz/MHz})^2} = 7.162 \times 10^3 \frac{G}{F^2} \]

Converting this into dB gives:

\[ A(\text{dBsm}) = 38.55 + G - 20 \log(F) \quad \text{with } F \text{ in MHz} \]

**G = −42.2 + 20 \log(D) + 20 \log(F) \quad \text{dB}**

Area of Isotropic Antenna = \(\frac{\lambda^2}{4\pi}\)

Area of Mouth of Dish Antenna = \(\frac{\pi D^2}{4}\)

The gain of the antenna is the ratio of the two areas = \(\frac{\pi^2 D^2}{\lambda^2} = \frac{\pi^2 F^2 D^2}{c^2}\)

To factor in 55% efficiency and allow input of \(F^2\) in MHz\(^2\), this must be multiplied by the factor \(0.55 \times 10^{12}\) (D is already in meters).

Combining this with the numerical values of \(\pi\) and \(c\) (both squared) makes the equation:

\[ G = \frac{(5.5 \times 10^{11})(3.14159)^2}{(3 \times 10^8)^2} D^2 F^2 = 6.03 \times 10^{-5} D^2 F^2 \]

which converts to dB form as:

\[ G = -42.2 + 20 \log(D) + 20 \log(F) \]

**Antenna Gain (not in dB) = 29,000/(\theta_1 \times \theta_2)**

for 55% efficient antenna
The area of an elliptical area on a sphere is:

\[
\frac{\pi^3 \ r^2 \ \theta_1 \ \theta_2}{(360)^2}
\]

where \(\theta_1\) and \(\theta_2\) are the two subtended angles (from the center of the sphere).

The area of the sphere is \(4\pi r^2\).

So the ratio of the surface of the sphere to the area included within the 3 dB bandwidth of an antenna with two mutually perpendicular beamwidths of \(\theta_1\) and \(\theta_2\) is:

\[
\frac{(720)^2}{\pi^2 \ \theta_1 \ \theta_2} = \frac{52,525}{\theta_1 \ \theta_2}
\]

Assuming that the antenna’s peak gain is equal to this area ratio (it’s pretty close), this would be the gain of a 100% efficient antenna. Then, multiplying by .55 would give the gain of a 55% efficient antenna, so:

Gain (not in dB) = \(\frac{28,889}{\theta_1 \ \theta_2} \approx \frac{29,000}{\theta_1 \ \theta_2}\)

**Noise Figure of Cascaded Stages**

The noise figure (F) determines the noise (N) (above kTB) that if added at the input of a noiseless device would cause the measured output noise. This would require N1 into the preamplifier and N2 into the receiver to cause the measured noise at the receiver output. If all noise were injected at the “input,” it would be (N2/G) + N1. This means that the system noise figure (at “input”) is degraded from F1 by:

\[
\text{Degradation} = \frac{N2/G + N1}{N1} = \frac{N2/G + N1}{N1} = \frac{N2}{G \times N1} + 1
\]

This degradation factor is converted to dB form and plotted in Figure 6-5 and in Appendix D.
Appendix B

Signal Strength in the Ether Waves

It is common practice (including throughout this book) to state the signal strength of transmitted signals in dBm, even though this makes no physical sense. dBm is a unit of electrical power, a ratio of the signal power to one milliwatt. Power is defined only within a circuit. After transmission from an antenna, signals are rigorously defined only in terms of field strength. The correct units are volts per meter (or more often microvolts per meter).

However, in many communication theory applications, it is extremely convenient to define a transmitted signal at some point in space in terms of dBm. That definition really assumes the situation shown in Figure B-1, in which an ideal unity gain antenna is located at the point in space being considered. The signal power in dBm is then the output of that ideal antenna at that location.

\[
P = -77 + 20 \log(E) - 20 \log(F)
\]

where,

- \(P\) = Equivalent signal power (dBm)
- \(E\) = Field strength (\(\mu\)V/m)
- \(F\) = Frequency (MHz)

Figure B-1  Electromagnetic field at an antenna

Quick Conversion Formulas

The following equation will allow you to convert field density in microvolts per meter directly to the equivalent signal strength in dBm.
To convert the signal strength back into the equivalent field density, use the formula:

\[ E = 10^{\left(\frac{P + 77 + 20 \log(F)}{20}\right)} \]

**Derivation**

In case you don’t believe these formulas, or have a particular need to derive something, here is the derivation of the first equation.

The signal strength (i.e., output power) from the ideal antenna in Figure B-1 is defined by the formula:

\[ P(\text{watts}) = \frac{[E(\text{v/m})]^2 A(\text{m}^2)}{Z_0(\text{ohms})} \]

where,

- \( P \) = Equivalent signal power
- \( E \) = Field strength
- \( A \) = Effective antenna area
- \( Z_0 \) = Impedance of free space

The effective antenna area can be defined as a function of antenna gain by the following formula:

\[ A = \frac{G \lambda^2}{2\pi} = \frac{G c^2}{4\pi F^2} \]

and the two constants are:

\[ Z_0 \approx 120 \pi \text{ ohms} \quad c = 3 \times 10^8 \text{ m/sec} \]

Setting the antenna gain equal to unity (i.e., 0 dB) and plugging the antenna area expression into the power equation gives:

\[ P = \frac{(E^2)(c^2)}{(480 \pi^2)(F^2)} \text{ volts}^2 \text{ meter}^2 \text{ meter}^2 \text{ sec}^2(1/\text{sec})^2 \text{ ohms} \]
\[ = 1.8998 \times 10^{13} \frac{E^2}{F^2} \text{ watts} \quad \text{(Combining all Units)} \]

But this gives the signal strength in watts and requires that the field density be input in volts/meter and the frequency in Hertz (not the most commonly used units). Multiplying the constant by the three factors:

\[ 10^{-12} \frac{\text{V}^2}{\mu \text{V}^2} \]
\[ 10^{-12} \frac{\text{MHz}^2}{\text{Hz}^2} \quad \text{(The frequency term is on the bottom)} \]
\[ 10^3 \frac{\text{mW}}{\text{W}} \]
yields the expression:

\[
P(\text{in mW}) = 1.8998 \times 10^{-8} \frac{E(\mu\text{V/m})^2}{F(\text{MHz})^2}
\]

which, when converted to dB form using the formulas in Chapter 2 and rounding the constant to the nearest whole number becomes:

\[
P = -77 + 20 \log(E) - 20 \log(F)
\]

(Q.E.D., as some may say.)
This appendix is a collection of formulas and nomographs, which will allow you to select the appropriate antenna for any type of communications application. All are explained in Chapter 4, Section 4.4.

In this appendix:

- \( G \) = Boresight gain (in dB unless otherwise noted)
- \( A \) = Effective area in \( m^2 \), \( \text{ft}^2 \), dBsm, or dBsf
- \( F \) = Frequency in MHz (or GHz on some charts)
- \( D \) = Antenna diameter in meters or feet
- \( \theta_1 \) & \( \theta_2 \) are the 3 dB beamwidths of an antenna in degrees in any mutually perpendicular planes

For a symmetrical parabolic dish antenna with 55% efficiency and diameter measured in meters:

\[
G = -42.2 + 20 \log(D) + 20 \log(F)
\]

If diameter is measured in feet, the formula is:

\[
G = -52.6 + 20 \log(D) + 20 \log(F)
\]

For a 55% efficient parabolic dish antenna:

\[
\text{Gain (not in dB)} \approx \frac{29,000}{\theta_1 \theta_2}
\]

For a 60% efficient horn antenna:

\[
\text{Gain (not in dB)} \approx \frac{31,000}{\theta_1 \theta_2}
\]

Effective area of an antenna as a function of gain and frequency is:

- \( A \) (in dBsm) = \( 38.6 + G - 20 \log(F) \)
- \( A \) (in dBsf) = \( 48.9 + G - 20 \log(F) \)
Table C-1  Antenna Selection Criteria

<table>
<thead>
<tr>
<th>Angular Coverage</th>
<th>Polarization</th>
<th>Bandwidth</th>
<th>Antenna Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>360° Azimuth</td>
<td>Linear</td>
<td>Narrow</td>
<td>Whip, Dipole, or Loop</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Wide</td>
<td>Biconical or Swastika</td>
</tr>
<tr>
<td>Circular</td>
<td>Narrow</td>
<td></td>
<td>Normal Mode Helix</td>
</tr>
<tr>
<td></td>
<td>Wide</td>
<td></td>
<td>Lindenblad or Four Arm Conical Spiral</td>
</tr>
<tr>
<td>Directional</td>
<td>Linear</td>
<td>Narrow</td>
<td>Yagi, Array with Dipole Elements or Dish with Horn Feed</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Wide</td>
<td>Log Periodic, Horn or Dish with Log Periodic Feed</td>
</tr>
<tr>
<td>Circular</td>
<td>Narrow</td>
<td></td>
<td>Axial Mode Helix or Horn with Polarizer or Dish with Crossed Dipole Feed</td>
</tr>
<tr>
<td></td>
<td>Wide</td>
<td></td>
<td>Cavity Backed Spiral, Conical Spiral or Dish with Spiral Feed</td>
</tr>
</tbody>
</table>

For 55% efficient antennas

![Graph](attachment:image.png)
Figure C-1  Antenna gain and area (diameter in meters)
Figure C-2  Antenna gain and area (diameter in feet)
Table C-2  Antenna Gain Adjustment vs. Efficiency

<table>
<thead>
<tr>
<th>Antenna Efficiency (%)</th>
<th>Adjustment to Gain from 55% Efficiency Value (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>60</td>
<td>Add 0.4</td>
</tr>
<tr>
<td>50</td>
<td>Subtract 0.4</td>
</tr>
<tr>
<td>45</td>
<td>Subtract 0.9</td>
</tr>
<tr>
<td>40</td>
<td>Subtract 1.4</td>
</tr>
<tr>
<td>35</td>
<td>Subtract 2</td>
</tr>
<tr>
<td>30</td>
<td>Subtract 2.6</td>
</tr>
</tbody>
</table>

Note: This table adjusts the gain calculated with the formulas on page 129.

Figure C-3  Antenna polarization

*Note: This applies between any linear and circular polarization combination
Appendix D
Quick Formulas for Receiver Sensitivity and Dynamic Range

This appendix is a collection of formulas and charts, which will allow you to calculate the sensitivity and dynamic range for a wide range of types of receivers and receiving systems. All are explained in Chapter 6.

Receiver Sensitivity (S) is the sum (in dB) of kTB, system noise figure (NF), and the required signal-to-noise ratio (SNR).

\[ S = kTB + NF + SNR \]

At standard room temperature (290°K):

\[ kTB = -114 \text{ dBm} + 10 \log(\text{Receiver Bandwidth}/1 \text{ MHz}) \]
Noise Figure in Cascaded Stages

For FM signals, the output SNR exceeds the RF SNR by the improvement factor:

$$\text{IF}_{\text{FM}} = 5 + 20 \log(\beta)$$

as long as the RF SNR $\geq 12$ dB for a conventional FM discriminator or 4 dB for a phase locked loop discriminator.
For digital signals, the signal-to-quantizing-noise ratio for a signal digitized with \( m \) bits is:

\[
SQR \text{ (in dB)} = 5 + 3(2^m - 1)
\]

<table>
<thead>
<tr>
<th>Bits/Sample</th>
<th>Quantizing Levels</th>
<th>Equivalent SNR (in dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>2</td>
<td>8</td>
</tr>
<tr>
<td>2</td>
<td>4</td>
<td>14</td>
</tr>
<tr>
<td>3</td>
<td>8</td>
<td>20</td>
</tr>
<tr>
<td>4</td>
<td>16</td>
<td>26</td>
</tr>
<tr>
<td>5</td>
<td>32</td>
<td>32</td>
</tr>
<tr>
<td>6</td>
<td>64</td>
<td>38</td>
</tr>
<tr>
<td>7</td>
<td>128</td>
<td>44</td>
</tr>
<tr>
<td>8</td>
<td>256</td>
<td>50</td>
</tr>
<tr>
<td>9</td>
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<td>56</td>
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<tr>
<td>10</td>
<td>1024</td>
<td>62</td>
</tr>
<tr>
<td>11</td>
<td>2048</td>
<td>68</td>
</tr>
<tr>
<td>12</td>
<td>4096</td>
<td>74</td>
</tr>
</tbody>
</table>

This chart as presented assumes that the signal is passed through an optimum matched filter for the modulation chosen. It is often drawn with the abscissa of the chart as \( \text{Eb/No in dB} \), which is the RFSNR adjusted for the bit rate to bandwidth ratio. The receiver system dynamic range is the difference between the sensitivity level and the signal strength of the strongest signal that can be present at the receiver input without causing spurious signals greater than the sensitivity level.
For the analog front end dynamic range, the following chart is used with values at the output of the preamplifier.

The dynamic range for a digital receiver is determined from:

$$DR = 20 \log_{10}(2^n)$$

### Dynamic Range vs. Digitizing Bits

<table>
<thead>
<tr>
<th>Number of Bits</th>
<th>Dynamic Range (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>4</td>
<td>24</td>
</tr>
<tr>
<td>5</td>
<td>30</td>
</tr>
<tr>
<td>6</td>
<td>36</td>
</tr>
<tr>
<td>7</td>
<td>42</td>
</tr>
<tr>
<td>8</td>
<td>48</td>
</tr>
<tr>
<td>9</td>
<td>54</td>
</tr>
<tr>
<td>10</td>
<td>60</td>
</tr>
<tr>
<td>11</td>
<td>66</td>
</tr>
<tr>
<td>12</td>
<td>72</td>
</tr>
<tr>
<td>13</td>
<td>78</td>
</tr>
<tr>
<td>14</td>
<td>84</td>
</tr>
<tr>
<td>15</td>
<td>90</td>
</tr>
<tr>
<td>16</td>
<td>96</td>
</tr>
</tbody>
</table>
Appendix E
Quick Formulas for Propagation

This appendix is a collection of formulas and nomographs, which will allow you to quickly calculate the signal strength at any point in a communication link—in terms of the common link parameters. For each formula or chart, you can select link distance in kilometers (km), statute miles (sm), or nautical miles (nm). All are explained in Chapters 5 and 7. Abbreviations used are:

- $G_T$ = Transmitting antenna gain (dB)
- $G_R$ = Receiving antenna gain (dB)
- $P_T$ = Transmitter power (dBm)
- ERP = Effective radiated power (dBm)
- $P_A$ = Power into receiving antenna (dBm)
- $P_R$ = Power into receiver (dBm)
- $L$ = All propagation losses (dB)
- $L_S$ = Atmospheric loss (dB)
- $L_{Atm}$ = Atmospheric loss (dB)
- $L_{Rain}$ = Loss from rain or fog (dB)
- $L_{LOS}$ = Loss from non-line-of-sight condition (dB)
- $F$ = Frequency (in MHz unless stated otherwise)
- $d$ = Distance from transmitter to receiver (various units)
- $H$ = Height (various units)
- $S$ = Sensitivity (dBm) [$S_{Rqd} = $ Required sensitivity]
- $M$ = Link margin (dB)

The Link Equation (several common forms)

- $P_R = P_T + G_T - L_S - L_{Atm} - L_{Rain} - L_{LOS} + G_R$
- $P_R = ERP - L_S - L_{Atm} - L_{Rain} - L_{LOS} + G_R$
- $P_R = ERP - L + G_R$
- $P_A = ERP - L$

Propagation Loss in dB is given by the following formulas:

For frequency in MHz and distance in kilometers (km):

- $L_S = 32 + 20 \log(F) + 20 \log(d)$
For frequency in MHz and distance in statute miles (sm):

\[ L_S = 37 + 20 \log(F) + 20 \log(d) \]

For frequency in MHz and distance in nautical miles (nm):

\[ L_S = 38 + 20 \log(F) + 20 \log(d) \]

**2-Ray Loss**

2-ray loss in dB is given by the following formulas:

For distance in kilometers and antenna heights in meters:

\[ L_{2RAY} = 120 + 40 \log(d) - 20 \log(h_T) - 20 \log(h_R) \]

For distance in statute miles and antenna heights in feet:

\[ L_{2RAY} = 148.9 + 40 \log(d) - 20 \log(h_T) - 20 \log(h_R) \]

For distance in nautical miles and antenna heights in feet:

\[ L_{2RAY} = 151.3 + 40 \log(d) - 20 \log(h_T) - 20 \log(h_R) \]

The following nomograph calculates the 2-ray loss for distance in kilometers and antenna heights in meters:
The following nomograph calculates 2-ray loss for distance in statute miles and antenna heights in feet:
The following nomogram calculates 2-ray loss for distance in nautical miles and antenna heights in feet:

**Fresnel Zone Distance**

Fresnel zone distance in kilometers from antenna heights in meters and frequency in MHz:

\[ FZ = \frac{(h_T \times h_R \times F)}{24,000} \]

Fresnel zone distance in statute miles from antenna heights in feet and frequency in MHz:

\[ FZ = \frac{(h_T \times h_R \times F)}{413,000} \]

Fresnel zone distance in nautical miles from antenna heights in feet and frequency in MHz:

\[ FZ = \frac{(h_T \times h_R \times F)}{478,000} \]
Atmospheric Loss for Various Distance Units

This is the same chart presented in Chapter 5; however, the loss in dB per statute mile (sm) or nautical mile (nm) can be found by extending the horizontal line out to the left hand scale.
Atmospheric Loss Through Whole Atmosphere

Knife edge propagation loss in the two nomographs on page 145 is based on the geometry shown below. Both show dB per unit distance vs. d.

\[ d = \sqrt{\frac{2}{1 + \frac{d_1}{d_2}}} d_1 \]
Knife Edge Diffraction Loss above $L_{FS}$

<table>
<thead>
<tr>
<th>H in meters</th>
<th>5</th>
<th>7</th>
</tr>
</thead>
<tbody>
<tr>
<td>4 k</td>
<td>4</td>
<td>8</td>
</tr>
<tr>
<td>2 k</td>
<td>3</td>
<td>9</td>
</tr>
<tr>
<td>1 k</td>
<td>2</td>
<td>10</td>
</tr>
<tr>
<td>600</td>
<td>1</td>
<td>12</td>
</tr>
<tr>
<td>400</td>
<td>0</td>
<td>16</td>
</tr>
<tr>
<td>200</td>
<td>1</td>
<td>18</td>
</tr>
<tr>
<td>100</td>
<td>0</td>
<td>20</td>
</tr>
<tr>
<td>60</td>
<td>25</td>
<td></td>
</tr>
<tr>
<td>40</td>
<td>30</td>
<td></td>
</tr>
<tr>
<td>20</td>
<td>35</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Freq in MHz</th>
<th>Path loss in dB (H above path)</th>
</tr>
</thead>
<tbody>
<tr>
<td>30</td>
<td>60</td>
</tr>
<tr>
<td>150</td>
<td>300</td>
</tr>
<tr>
<td>600</td>
<td>1,500</td>
</tr>
<tr>
<td>3,000</td>
<td>6,000</td>
</tr>
<tr>
<td>15,000</td>
<td>30,000</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>H in feet</th>
<th>5</th>
<th>7</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.6</td>
<td>0.1</td>
<td>0.6</td>
</tr>
<tr>
<td>0.4</td>
<td>0.4</td>
<td>0.4</td>
</tr>
<tr>
<td>0.2</td>
<td>0.2</td>
<td>0.2</td>
</tr>
<tr>
<td>0.1</td>
<td>0.1</td>
<td>0.1</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Path loss in dB (H above path)</th>
</tr>
</thead>
<tbody>
<tr>
<td>30</td>
</tr>
<tr>
<td>35</td>
</tr>
</tbody>
</table>

Quick Formulas for Propagation

143
Additional Loss above $L_{FS}$ from Curvature of Smooth Earth

![Graph showing dB loss for curvature of earth with frequency (MHz) and distance in miles as variables.](image-url)
Spreading Loss ($L_{FS}$) as a Function of Distance and Frequency
Attenuation from Rain or Fog

The attenuation lines on the charts on pages 148 and 149 are all defined by the tables at the bottom of this page.

<table>
<thead>
<tr>
<th>Rain</th>
<th>Frequency—GHz</th>
<th>$L_{\text{Rain}}$ in dB per kilometer</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>0.25 mm/hr</td>
<td>.01 in/hr</td>
</tr>
<tr>
<td>B</td>
<td>1.0 mm/hr</td>
<td>.04 in/hr</td>
</tr>
<tr>
<td>C</td>
<td>4.0 mm/hr</td>
<td>.16 in/hr</td>
</tr>
<tr>
<td>D</td>
<td>16 mm/hr</td>
<td>.64 in/hr</td>
</tr>
<tr>
<td>E</td>
<td>100 mm/hr</td>
<td>4.0 in/hr</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Fog</th>
<th>Visibility</th>
</tr>
</thead>
<tbody>
<tr>
<td>F</td>
<td>greater than 600 m</td>
</tr>
<tr>
<td>G</td>
<td>about 120 m</td>
</tr>
<tr>
<td>H</td>
<td>about 30 m</td>
</tr>
</tbody>
</table>
Quick Formulas for Propagation

\[ L_{\text{Rain}} \text{ in dB per statute mile} \]

Frequency—GHz

\[ L_{\text{Rain}} \text{ in dB per nautical mile} \]

Frequency—GHz
Link Relationship Formulas

Link Margin

\[ M = P_R - S \]

Required Sensitivity

\[ S_{Rqd} = ERP - 32 - 20 \log(F) - 20 \log(d) \]
\[ - L_{Atm} + G_R - M \quad (\text{for Distance in km}) \]
\[ S_{Rqd} = ERP - 37 - 20 \log(F) - 20 \log(d) \]
\[ - L_{Atm} + G_R - M \quad (\text{for Distance in sm}) \]
\[ S_{Rqd} = ERP - 38 - 20 \log(F) - 20 \log(d) \]
\[ - L_{Atm} + G_R - M \quad (\text{for Distance in nm}) \]

Effective Range

\[ d = 10^{\frac{20 \log(d)}{20}} \]

where,

\[ 20 \log(d) = ERP - S + G_R - L_{Atm} - M - 32 - 20 \log(F) \quad \text{for } d \text{ in km} \]
\[ 20 \log(d) = ERP - S + G_R - L_{Atm} - M - 37 - 20 \log(F) \quad \text{for } d \text{ in sm} \]
\[ 20 \log(d) = ERP - S + G_R - L_{Atm} - M - 38 - 20 \log(F) \quad \text{for } d \text{ in nm} \]

Interfering Signal to Desired Signal Ratio

\[ I/S = ERP_I - ERP_S - L_I + L_S - I_A - I_F \]

\((I_A \text{ is the antenna isolation in dB and } I_F \text{ is the frequency filtering isolation in dB})\)

For in band interference with negligible difference in atmospheric attenuation

\[ I/S = ERP_I - ERP_S - 20 \log(d_I) + 20 \log(d_S) - I_A \]

For out of band interference with negligible difference in atmospheric attenuation

\[ I/S = ERP_I - ERP_S - 20 \log(d_I) + 20 \log(d_S) \]
\[ - 20 \log(F_I) + 20 \log(F_S) - I_A - I_F \]
Appendix F
Quick Formulas for Doppler Shift

Where:

- $\Delta F = F_R - F_T = \frac{V}{C} F_T$
- $F_T =$ The transmitted frequency
- $F_R =$ The received frequency
- $V =$ The received velocity (toward each other)
- $C =$ The speed of light

*Figure F-1  Doppler shift*

\[ \Delta F = \frac{F_T}{C} \left( V_T \cos(\theta_T) + V_R \cos(\theta_R) \right) \]

*Figure F-2  Doppler effect for arbitrary velocity vectors*
The slide rule has scales for calculation of antenna parameters, free space propagation, 2-ray propagation, the Fresnel zone distance and dB conversion. Figure G-1 shows the location of each set of scales on the rule. The accuracy of calculations using this slide rule is approximately 0.2 dB.

**Antenna Scales**

To find antenna gain and beamwidth from antenna diameter, operating frequency and antenna efficiency. Set antenna diameter to operating frequency as in Figure G-2. First, move the slide so that the frequency is next to the antenna diameter on the top window of side 1.

Now move down to the second window and read the bore sight gain against the efficiency as in Figure G-3.

Now look at the beamwidth scale at the top of the second window on side as shown in Figure G-4. You can read the 3 dB and 10 dB beamwidths and the angle from bore sight to the first null and the first side lobe maximum.

Now set the frequency at the third window on side 1 and look at the bottom window on side 1, as in Figure G-5. The gain as calculated above must be reduced by the number of dB next to the surface tolerance.

**Free Space Propagation Loss**

This is for loss between two isotropic antennas. Look at the center part of side 1 of the slide rule. Move the slide so that the transmit frequency is by the arrow at point A on Figure G-6. Then, read the free space loss in dB against the link distance in km at point B and the short range free space loss at point C on Figure G-7.

**2-Ray Propagation Loss**

This is for the attenuation between two isotropic antennas. Move the slide until the transmit antenna height is at the link distance at point A in the first window on side 2 as shown in Figure G-8. Read the 2-ray attenuation next to the operating frequency at point B in the same window.
Figure G-1  Antenna and propagation calculator with the function of each scale identified
Figure G-2  Antenna diameter aligned with frequency
Figure G-3  Gain vs. efficiency

Figure G-4  3 dB beamwidth, 10 dB beamwidth, 1st sidelobe-max and 1st null

Figure G-5  Gain reduction vs. surface tolerance
Fresnel Zone Distance

This calculates the range at which the propagation mode changes from free space to 2-ray in line-of-sight conditions. Move the slide so that the transmit antenna height is aligned with the receiving antenna height as shown at point A in Figure G-9. Then read the Fresnel zone distance opposite the operating frequency at point B in the same window.
Conversion to dB

This scale converts a power ratio to dB. Move the slide so that the power ratio is at the arrow in the bottom window of side 2 as shown in Figure G-10. Read the ratio in dB at the bottom arrow in the same window.

Figure G-9  Fresnel zone calculation scales

Figure G-10  dB conversion
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