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Zerihun Abate

WiMAX RF SYSTEMS ENGINEERING

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WiMAX RF Systems Engineering

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To my mother, Gete Ayele,
whose love and prayers sustained me through the years,
and to my late father, Abate Negatu,
who brought me up with love and instilled confidence in me.

Contents

Preface

We have heard the adage, “Need taught him wit,” and others similar to it such as, “Necessity is the mother of invention.” The saying is no more true anywhere else than in telecommunications.

As societal needs and demands change, they fuel technology to rise and meet those needs. (It can be argued that sometimes the antithesis is true, as advances in technology can create “unnecessary” needs.) Largely, though, needs give a strong impetus to technology to innovate. As society becomes more complex, problems abound, food becomes scarcer, health issues arise, people spread out in space and time, and innovation and efficiency requirements take center stage.

Communication as a crucial part of technology has evolved very rapidly as a result. The wireless communication technology, namely, wireless mobile communication in particular, has evolved over a few generational changes in a short period that represent applications from the crude to the advanced.

Here is a summary of the wireless mobile communication generational changes:

- *0G:* Prior to first generation (1G), mobile phone was limited to two-way radios such as in police cruisers, taxicabs, and ambulances. In the early 1940s Motorola developed backpacked two-way radio and a walkie-talkie and later developed a large, handheld two-way radio for the military.

- *1G:* Mobile communication based on analog cellular technology was first launched in 1979 (in Japan) and 1981 (in the Scandinavian countries). An example of 1G system in the United States is the Advanced Mobile Phone Systems (AMPS).
- *2G:* A digital version of mobile communication came to the fore in the 1990s. The 1G system includes: GSM, IS-136 (TDMA), iDEN,[1] and IS-95 (CDMA). The various improvements and evolutions in each of these systems gave rise to a change of name to distinguish them from the basic 2G as in 2.5G, 2.75G, and just short of 3G.
- *3G:* 3G services are designed to offer broadband cellular access at speeds of 2 Mbps, which will allow multimedia services. In 1998 the International Telecommunication Union (ITU) called for and accepted a proposal for IMT-2000[2] (3G) from different entities based on TDMA and CDMA technologies. The European Telecommunication Standards Institute (ETSI) and Global System for Mobile Communication (GSMC) with infrastructure vendors such as Nokia and Ericsson backed WCDMA, while the U.S. vendors, Qualcomm and Lucent Technologies, backed CDMA2000. Improvements to accommodate a higher data rate over the basic 3G has taken place, and 3G currently includes 1x, EVDO; WCDMA, and HSPA. In this evolution, at the time of this writing, we are now in transition between 3G and 4G. Incidentally, this is where WiMAX injects itself in that continuance.
- *4G:* 4G is a convergence point for various CDMA technology flavors, at least in terms of access technology, to OFDMA, and as such these latest and up-and-coming technologies include: IEEE 802.16e (WiMAX), IEEE 802.16m, long-term evolution (LTE), and LTE-Advanced.

While the definition for fourth generation (4G) is somewhat imprecise, it can include many of the following features:

- Services include interactive, multimedia, voice, and video stream;

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[1] Operating in 800 MHz and 900 MHz using a 25-kHz channel bandwidth on TDMA/GSM architecture, iDEN (Integrated Digital Enhanced Network) combines the capabilities of a digital cellular telephone, a two-way radio, an alphanumeric pager, and a data/fax modem in a single network while giving the end user access to information without having to carry around several devices.

[2] IMT-2000 (International Mobile Telecommunications) is the term used by the ITU for a set of globally harmonized standards for 3G mobile telecom services and equipment that provide higher data speed between mobile phones and base antennas. IMT-Advanced is for the generation beyond 3G.

- Replaces the core network (CN) with 100% IP-based network;
- Much higher throughput and data rate than 3G (100 Mbps to 1 Gbps);
- Higher spectrum efficiency (8 bits/sec/Hz);
- OFDMA technology for higher data rate and immunity to ISI;
- Mobility services;
- Lower latency;
- Better security
- Enhanced QoS;
- Usage of MIMO antenna technology for capacity and throughput;
- Usage of multiple or concatenated coding which improves bit error rate in FEC and multiple QoS capability by adding redundancy (over and above, say, convolutional coding).

Book Organization

In the pages of this book, you will find fundamental WiMAX RF network planning principles laid out step by step and supported by concrete examples. Without getting bogged down with too much detail and unnecessary mathematical rigor, the essentials are presented in a straightforward manner. Not only the whys, but also the hows are explained.

Even though this book is titled *WiMAX RF Systems Engineering,* the engineering principles and methods explained extend to most other similar wireless technology fields.

As a more hands-on than a theoretical text, this book is suitable for design engineers, experienced technicians, or communications students. It begins with a general historical background and introductory information on WiMAX technology. A case is made why WiMAX is a compelling technology, and the book covers spectrum definition; channel characterization; WiMAX modulation and antenna techniques; link budgets; coverage, capacity, and frequency planning; service delays; interference; intermodulation and noise description; illustrations; and supporting examples.

The book is divided into five major parts, as shown in Figure P.1. It begins with the historical and technical background of WiMAX, which makes up Parts I and II. Part III relays the core message of the book, dealing with network radio design and covering such topics as link design, link budget, point-to-point link design, network planning steps, WiMAX capacity, frequency plan, antennas, oversubscription, delay characterization, case study, and benchmarking.

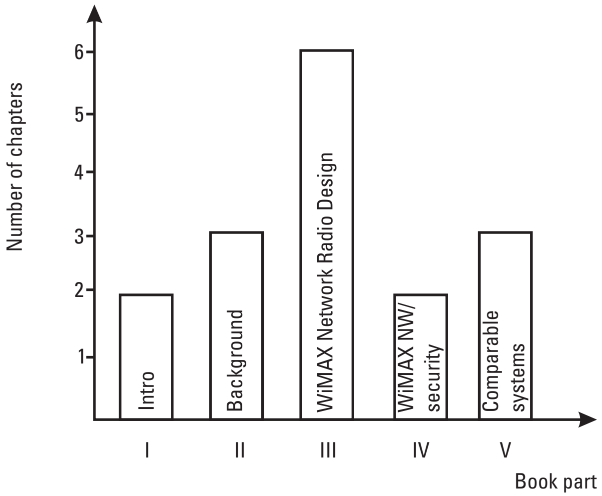


Figure P.1 Book organization.

Part IV covers network and security aspects of WiMAX, while Part V deals with systems comparable to WiMAX, such as WiFi radio design and UMTS.

Part I
Introduction

1

Introduction

In what is to be considered a major technology milestone, in September 2008, XOHM launched Mobile WiMAX in a challenging RF propagation environment in Baltimore, Maryland. Sprint's XOHM and its partners claim that they have covered 70% of the city and will continue to build the technology throughtout the rest of the United States [1].

Worldwide interoperability microwave access (WiMAX) is a burgeoning IP-based wireless technology that has already begun to arrive. As ubiquitous wireless services to fixed, nomadic or mobile subscribers that can deliver data rates rivaling or exceeding cable and digital subscriber line (DSL), WiMAX offers easily realizable promises to deliver voice, data, and video signals over the Internet throughout the world, including rural communities of emerging nations.

Analogous to wireless local loop (WLL), in which a subscriber is offered a landline-type phone service delivered wirelessly from housetop residential building antennas, WiMAX has many more applications than just voice service. WLL, in its limited application, proved advantageous, especially for developing nations where a telecommunication infrastructure is not widely available and routing wires and deploying a landline system proved cost prohibitive and time consuming. In a similar fashion, WiMAX can be widely deployed in an economical fashion.

Not based on absolutely brand new technology, WiMAX has had forerunners, such as wireless Internet service provider (WISP) in the late 1990s, albeit with limited success. Another antecedent bearing closer similarity to WiMAX was local multipoint distribution systems (LMDS), a short lived service targeting business users and operating in the 24- and 39-GHz bands. LMDS was doomed on account of its mainly small cell radius resulting from its high operating frequency.

Similarly, another precursor technology to WiMAX that came to the fore was multichannel multipoint distribution systems (MMDS). Initially targeting rural communities to provide wireless cable broadcasting and later wireless Internet service, MMDS operated in the 2.5-GHz range. To its credit, MMDS provided coverage of over 30 miles because of its low operating frequency coupled with tall antenna heights used that enabled more line-of-site (LOS) access. Despite and because of its large coverage areas, capacity was limited. Elsewhere in the world similar LOS systems were deployed in 3.5 GHz.

While LMDS and MMDS LOS performances were good, the nonline-of-sight- (NLOS-) related multipaths and the attendant intersymbol interference (ISI) issues left a lot to be desired. However, from a data transmission standpoint, it is easy to see that MMDS bore the closest resemblance to WiMAX technology, the subject of this book. In the rest of this book, we will see how WiMAX addresses the shortcomings of its precursor technologies and capitalizes on its own strengths.

1.1 WiMAX System

Designed for broadband wireless access (BWA) and operating above 2 GHz, WiMAX is based on an RF technology called orthogonal frequency division multiplexing (OFDM) and is an effective means of transferring high-speed data, especially when the bandwidth used is 5 MHz or greater.

WiMAX wireless access comprises a WiMAX base station (BS) and customer premise equipment (CPE) for indoor service, and a mobile unit or modem for outdoor service. WiMAX BS theoretically covers a range of up to 30 miles, but it is really limited to a more practical range of 6 miles. Data rates supported ranges of up to 70 Mbps.

WiMAX backhaul, interconnecting BSs, is a line-of-sight (LOS) service, while subscriber station (SS) and BS can be either a LOS or a nonline-of-sight (NLOS) service. Also, in mobile WiMAX, the SS can be in motion. [Throughout this book, *subscriber station* (SS) is interchangeably used with *mobile station* (MS) or *handset*. *Base station* (BS) and *cell site* may also be used interchangeably.]

As shown in Figure 1.1, the basic applications and services offered by WiMAX are:

1. Fixed—For the home Internet user, in place of or complement to cable or DSL;
2. Phone—For home phone service via Voice-over-Internet Protocol (VoIP), including video calls;

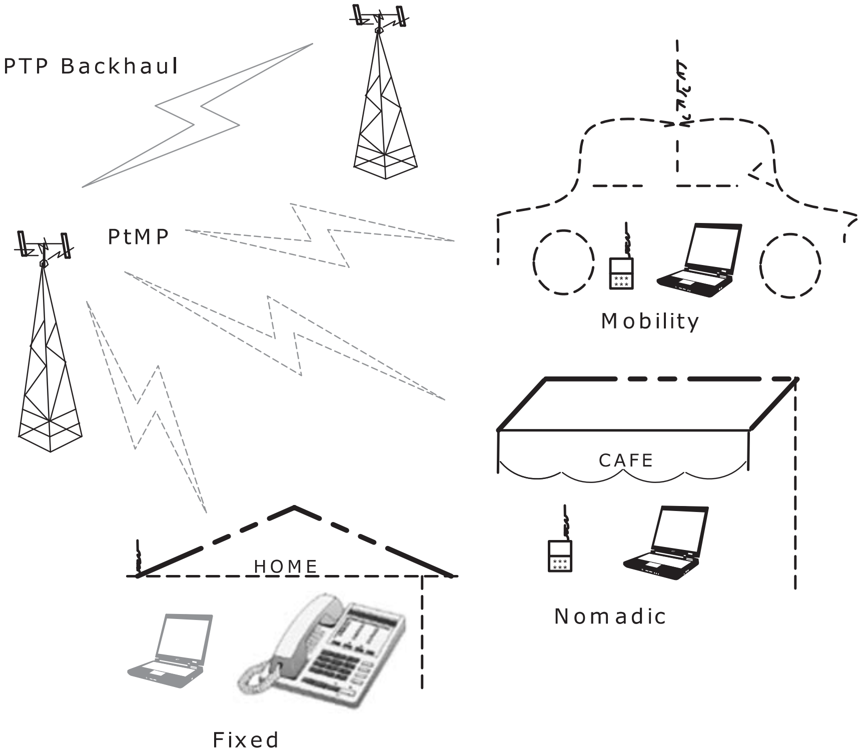


Figure 1.1 WiMAX system and application.

3. Nomadic—For laptop Internet users at various locations where service may be available, for example, in cafés or libraries;
4. Mobility—For laptop/handset users for Internet data access or just for handset users for a VoIP application. (*Mobility* implies pedestrian or vehicular use.)

The outstanding advantages of WiMAX include:

- An all-IP, all-packet technology;
- Low cost;
- An IP packet-based system that benefits from computer equipment advances;
- WiMAX is concerned with layers 1 and 2 (physical and MAC layers) and does not define connectivity at the network layer (layer 3), leaving third parties to innovate and standardize at the higher layers;

- A connection to the legacy systems of wireline carriers' IP cores or wireless operators' IP cores;
- An IP multimedia subsystem (IMS) providing Internetworking roaming;
- Compatibility with 3G cellular;
- IP-based QoS and common application;
- Leveraging the investment already made in the existing core network;
- an alternative to laying copper lines for DSL/cable operators, while using their existing IP core network;
- Service for hard-to-reach areas/rural areas.

1.2 WiMAX and Related Standards

The working group of IEEE 802.16 developed the standard of WiMAX, which is also called wireless area metropolitan networks (WMANs). The WiMAX-technology-based IEEE 802.16 standard also addresses the European Telecommunications Standards Institute's (ETSI's) high performance radio metropolitan area network (HiperMAN) standard, rendering it a worldwide compatible standard.

- The original 802.16 standard specified transmissions in the range of 10–66 GHz;
- 802.16d included lower frequencies in the range of 2–11 GHz;
- Frequencies commonly used for 802.16d are 3.5 and 5.8 GHz;
- Frequencies commonly used for 802.16e are 2.3, 2.5, and 3.5 GHz, depending on the country.

Refer to Figure 1.2, which shows the WiMAX band allocation for now and the future. Note that 802.16d or earlier versions are for fixed applications, while 802.16e is for mobile applications [2].

IEEE is also currently working on a new wireless standard called 802.16m. 802.16m is an OFDMA-compatible new high speed standard with a promise of cross platform compatibility with WiMAX and 4G. More details about 802.16m follow.

1.2.1 WiMAX 802.16d (802.16-2004)

WiMAX 802.16d provides:

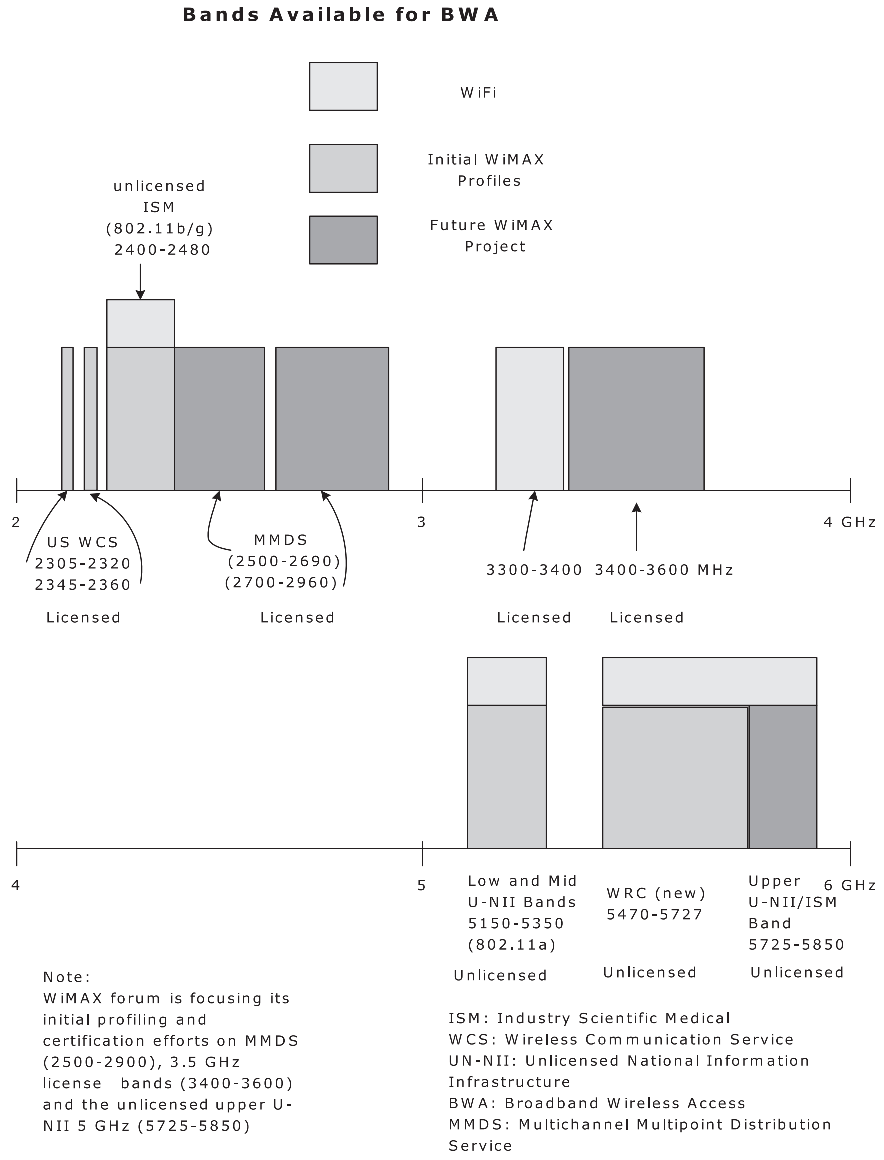


Figure 1.2 WiMAX and WiFi RF spectra. (*From:* [2]. © 2004 Fujitsu Microelectronics America, Inc. Reprinted with permission.)

- Last-mile access to residences for basic telephony services in the absence of a wireline infrastructure, such as in rural areas of developing countries;
- Wireless Internet service;

- Backhaul, interconnecting cell sites;
- Temporary backhauls for sporting events and trade shows.

1.2.2 WiMAX 802.16e (802.16-2005)

WiMAX 802.16e provides:

- Nomadic and mobile telephony using VoIP;
- Nomadic and mobile multimedia based services on Internet protocol (IP).

1.2.3 WiMAX 802.16/d/e and Related Standards Summary

See Table 1.1 for a summary of WiMAX 802.16/d/e and related standards specifications.

See Table 1.2 for descriptions of the IEEE 802.16 evolution of family of standards relevant to WiMAX.

1.2.4 Upcoming 802.16m Standard

IEEE is currently working on a new wireless standard called 802.16m [4].

802.16m is a new OFDMA-compatible, high-speed standard with a promise of cross-platform compatibility with WiMAX. 802.16m will also be

Table 1.1
WiMAX 802.16/d/e Standards Summary

	802.16	**802.16d/HiperMAN**	**802.16e**
Completed	December 2001	June 2004 (802.16d)	2005
Spectrum	10–66 GHz	< 11 GHz	< 6 GHz
Channel Conditions	Line-of-sight service only	Nonline-of-sight service	Nonline-of-sight service
Bit Rate	32–134 Mbps in 28-MHz channel bandwidth	Up to 75 Mbps in 20-MHz channel bandwidth	Up to 15 Mbps in 5-MHz channel bandwidth
Modulation	QPSK, 16 QAM and 64 QAM	OFDM 256 FFT QPSK, 16 QAM, 64 QAM	Scalable OFDMA 128–2,048 FFT/BPSK, QPSK, 16 QAM, 64 QAM
Mobility	Fixed	Fixed	Nomadic/mobile
Channel Bandwidths	20, 25, and 28 MHz	1.75–20 MHz	1.75–20 MHz

Table 1.2
IEEE 802.16 Evolution of Standards Relevant to WiMAX

Standards/ Amendment	Comments
802.16	Now withdrawn. This is the basic 802.16 standard that was released in 2001. It provided for basic high data links at frequencies between 11 and 60 GHz.
802.16a	Now withdrawn. This amendment addressed certain spectrum issues and enabled the standard to be used at frequencies below the 11 GHz minimum of the original standard.
802.16b	Now withdrawn. It increased the spectrum that was specified to include frequencies between 5 and 6 GHz while also providing for Quality of Service aspects.
802.16c	Now withdrawn. This amendment to 802.16 provided a system profile for operating between 10 and 66 GHz and provided more details for operations within this range. The aim was to enable greater levels of interoperability.
802.16d (802.16-2004)	This amendment was also known as 802.16-2004 in view of the fact that it was released in 2004. It was a major revision of the 802.16 standard and upon its release, all previous documents were withdrawn. The standard / amendment provided a number of fixes and improvements to 802.16a including the use of 256 carrier OFDM. Profiles for compliance testing are also provided, and the standard was aligned with the ETSI HiperMAN standard to allow for global deployment. The standard only addressed fixed operation.
802.16e (802.16-2005)	This standard, also known as 802.16-2005 in view of its release date, provided for nomadic and mobile use. With lower data rates of 15 Mbps, against 70 Mbps in 802.16d, it enabled full nomadic and mobile use including handover.
802.16f	Management information base
802.16g	Management plane procedures and services
802.16h	Improved coexistence mechanisms for license-exempt operation
802.16j	Multi-hop relay specification
802.16k	802.16 bridging
802.16Rev2	Amendment under development: Aims to consolidate 802.16-2004, 802.16e, 802.16f/g/i into new document
802.16m	Advanced air interface. This amendment is looking to the future and it is anticipated it will provide data rates of 100 Mbps for mobile applications and 1 Gbps for fixed applications. It will allow cellular, macro and micro cell coverage, with currently there are no restrictions on the RF bandwidth although it is expected to be 20 MHz or more.
802.20	Mobile broadband wireless access, similar to 802.16e. Both 802.16e and 802.20 address the new mobile air interface, however, 802.16e is based on existing 802.16 standards whereas 802.20 is a standalone standard being developed from scratch. An additional difference is 802.16e specifies mobility in the 2-6 GHz range, whereas 802.20 targets operation in licensed bands below 3.5 GHz.

Source: from [3]

compatible with the future 4G. When 4G is implemented all systems, including the WCDMA and CDMA2000 standards will be OFDMA compatible.

IEEE 802.16m amends IEEE 802.16 Wireless MAN-OFDMA specification to provide an advance air interface for operation in licensed bands. It will meet the cellular-layer requirements of IMT–Advanced (international mobile telecommunication) next generation mobile networks and will be designed to provide significantly improved performance compared to other high-rate broadband cellular network systems.

Currently 802.16m specifications include the following:

- Very low data rate—16 Kbps;
- Low rate data and low multimedia—144 Kbps;
- High multimedia—2 Mbps;
- Super-high multimedia—30–100 Mbps (mobile)/1 Gbps (fixed).

The uplink speeds are still not determined.

802.16m seeks to solve the handover issues of 802.16e, namely, problems with expedited ranging prior to handover, as well as a possible lack of acknowledgment to the serving BS when switching to the anchor BS under low uplink signal-to-ratio conditions.

The capacity improvement behind 802.16m is the use of MIMO antennas.

The expected completion for 802.16m is December 2009.

1.3 Range and Spectrum Efficiency Summary

Table 1.3 shows a range comparison of WiMAX and WiFi, while Table 1.4 shows spectrum efficiency comparison of the same. Figure 1.2 shows the available bands for WiMAX and WiFi.

Table 1.3

Range Comparison

	Fixed WiMAX	**Mobile WiMAX**	**WiFi**
Frequency*	3.5 GHz, 5.8 GHz	2.3 GHz, 2.5 GHz	2.4 GHz, 5 GHz
Coverage (Typical Radius)	3–5 miles	< 2 miles	< 100 feet (indoor/outdoor)

*2.4 GHz and 5.8 GHz are typically used for WiFi and cordless telephone.

Table 1.4
Spectrum Efficiency Comparison

WLAN (WiFi)	WiMAX
2.7 bps/Hz (54 Mbps in 20-MHz channels)	5 bps/Hz (100 Mbps in 20-MHz channels)
No quality of service (QoS)	MAC QoS supports service differentiation levels.
Up to 100m coverage	12–15 km (LOS) 1–2 km (NLOS)

Even though WiMAX supports frequencies in the range of 2–11 GHz, the deployment focus from a global perspective is in the frequencies of 2.3 GHz, 2.5 GHz, 3.5 GHz, and 5.7 GHz.

1.4 WiMAX Spectrum Worldwide

Table 1.5 shows WiMAX spectrum allocations around the world.

1.5 What Is OFDM?

Orthogonal frequency division multiplex (OFDM) is a multicarrier modulation scheme in which, just as for frequency division multiple access (FDMA), a channel is subdivided into subchannels or subcarriers. Only in OFDM the high data rate message to be transmitted is converted to a number of parallel slower bitstreams, each of which modulates a subcarrier. The subcarriers are then transmitted at the same time. Figures 1.3 and 1.4 illustrate the principles and concepts of OFDMA.

1.6 The Key Features of WiMAX

WiMAX has numerous advantageous features mostly centered around its modulation, subchanneling, and coding schemes. The salient ones are outlined in the following sections.

Table 1.5
Major Spectrum Allocations for WiMAX Worldwide

Region	Frequency Bands (GHz)	Comments
Canada	2.3	
	2.5	
	3.5	
	5.8	
United States	2.3	
	2.5	
	5.8	
Central and South America	2.5	Spectrum fragmented and varies from country to country
	3.5	
	5.8	
Europe	2.5	Spectrum fragmented and varies from country to country
	3.5	
	5.8	
Middle East and Africa	2.5, 5.8	Spectrum fragmented
Russia	2.5, 3.5, 5.8	The 2.5-GHz allocation is currently allocated to IMT 2000
Asia Pacific (including China, India, Australia, and so on)	2.3	Spectrum fragmented and varies from country to country
	3.3	
	3.5	
	5.8	

Source: [3].

1.6.1 Intersymbol Interference (ISI)

The chief advantage of the OFDM waveform is that the symbol duration of the subcarriers is increased in relation to the delay spread, which makes OFDM highly tolerant to multipath-caused intersymbol interference (ISI) [5].

Digital transmission systems' high-speed symbols modulating a carrier, as shown in Figure 1.3, are spaced closely in time and can cause ISI at the receiver.

This is best explained through an example:

Let the high-speed data rate, R_d = 5 Mbps, correspond to a single carrier.

Then the symbol period, $T_S = \dfrac{1}{R_d} = 0.2\mu s.$

Let the delay spread of the channel (reflected symbols' arrival time), T_d = 60 μs.

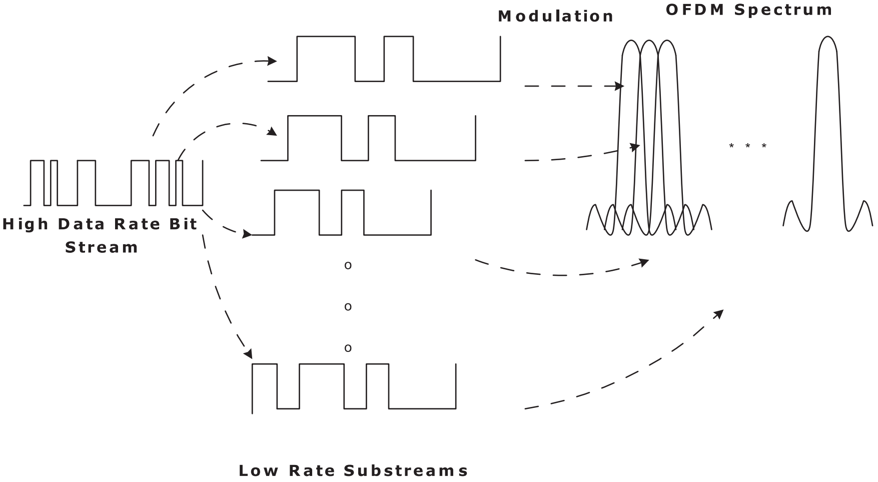


Figure 1.3 OFDM concept.

During the 60-μs delay, there are 60/0.2, or 300, symbols.

That means the delayed symbols arriving within the 60 μs will interfere with the next 300 symbols.

However, let's assume a multicarrier system with 512 subcarriers.

Now the low-stream data rate in each subcarrier is 5 Mbps/512 = 9.76 kbps.

The symbol rate for this is, $T_s = \dfrac{1}{9.76 \text{ kbps}} = 102\ \mu s.$

Now the 60-μs delayed symbols are within the symbol rate of 102 μs, and hence, they cannot cause ISI.

It is instructive to mention that the Doppler effect, in mobile SS, has the opposite effect of multipath by surpassing or exceeding, so to speak, the coherence bandwidth or, in the time domain, reducing the coherence time. Hence, for mobile WiMAX, a compromise is needed to accommodate both multipath and Doppler effects.

1.6.2 Frequency Selective Fading

One of the crucial advantages of OFDM is a mitigation of frequency-selective fading. Frequency-selective fading can be very severe in a multipath channel, especially for a channel with no Rician (direct path) and only Rayleigh distribution. The fading due to multipath can be on the order of tens of decibels, on account of a low signal-to-noise ratio (SNR) under these conditions. However, in the case of OFDM modulation, the data is distributed among multiple

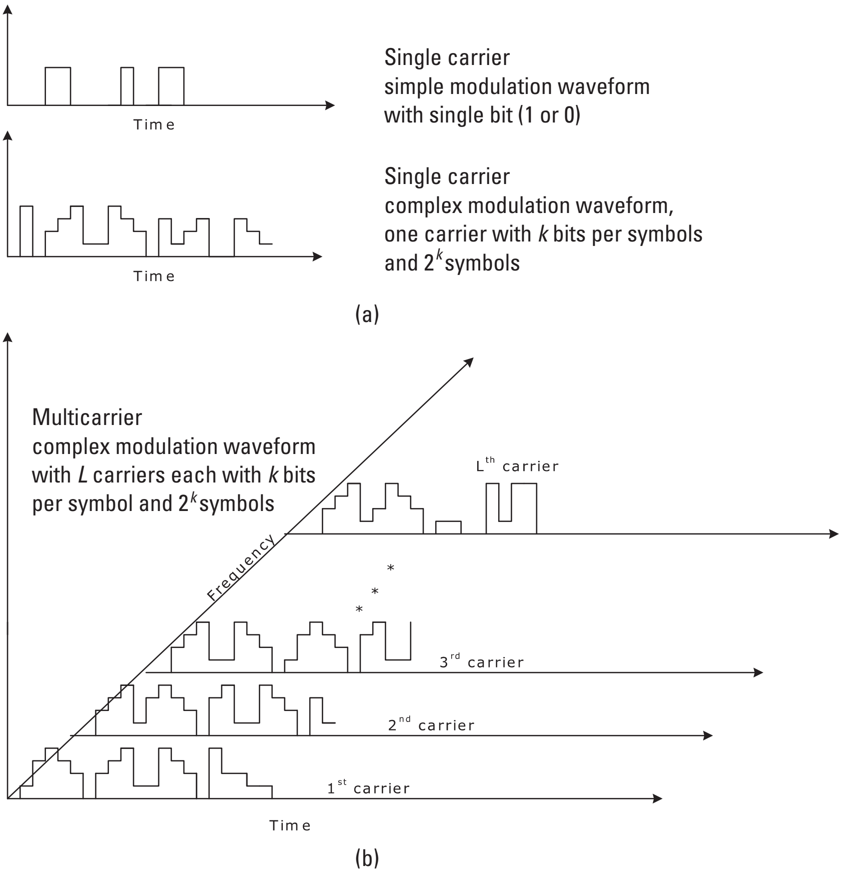


Figure 1.4 OFDM multicarrier concept: (a) single carrier and (b) multicarrier. (*After:* [6].)

subcarriers, and the impacted information in a few subcarriers can be recovered using error-correction coding.

1.6.3 Adaptive Modulation and Coding (AMC)

On a per-frame basis, WiMAX is able to change the forward error correction (FEC) and modulation types depending on the channel environment, thereby optimizing throughput.

1.6.4 Very High Data Rate

When operating at a 20-MHz bandwidth, 5/6 FEC, and 64 quadrature amplitude modulation (QAM), up to a 73-Mbps data rate can be achieved.

The aforementioned benefits are not without cost. More particularly, WiMAX is subject to a reduction of throughput in an NLOS environment and adverse weather conditions, such as rain, especially for higher bands in fixed point-to-point or LOS applications.

1.6.5 Delay Versus Doppler Spreads and Frequency Diversity

In orthogonal frequency division multiple access (OFDMA), applicable to mobile WiMAX, FFT size is scalable from 128 to 2,048, but the carrier spacing is kept constant at 10.94 kHz. This keeps the symbol duration, the inverse of carrier spacing, constant. The 10.94 kHz was chosen as a tradeoff between delay spread and Doppler spread, which normally require an opposite compensation to mitigate.

OFDMA also allows subchannelization, or grouping of subcarriers, in both uplink and downlink, which provides frequency diversity.

1.7 WiMAX Transceiver

Figure 1.5 shows a WiMAX transmitter and receiver block diagram. In the transmitter, the serial-to-parallel converted data is mapped in the modulator. As a centerpiece of the WiMAX transmitter, the inverse fast Fourier transform (IFFT) (FFT in the receiver) realizes the time domain WiMAX signal by modulating each data symbol onto a unique carrier frequency. The remaining steps are parallel-to-serial and digital-to-analog conversions followed by RF modulation for transmission. The reverse is done in the receiver.

1.8 OFDM Signal Generation

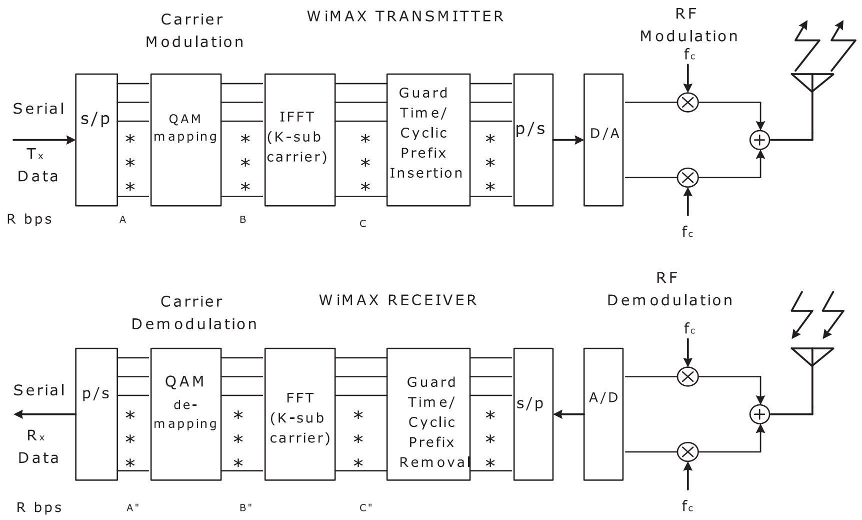
Refer to Figure 1.6, which shows the OFDM signal realization.

Equations (1.1) and (1.2) [6] show the derivation of how OFDM signals can be realized using IFFT. IFFT of a signal a_k is given by:

$$a(n) = \sum_{k=0}^{N-1} a_k e^{\frac{j2\pi kn}{N}} \qquad n = 0 \ldots N-1 \tag{1.1}$$

$a(n)$ is a sampled function where $\vec{a}_K$ is complex and $1 \le k \le N$ denotes the successive bits stored, $n = 0\ldots N-1$.

Equivalent to the inverse Fourier transform, (1.1) can be used to realize OFDM. The next step would be to RF-modulate the signal for transmission as:



A and B
Using S/P and QAM mapper, K symbols are generated.
Each of R/K rate and bandwidth (BW) of BW/K. K is the number of sub-carriers (in 10 MHz BW there are 1024 subcarriers).

B
Here data is coded in frequency domain, one symbol at a time

C
Data in time doman, one symbol at a time. FFT is useful to to have a tight orthogonality b/n the carriers while modulating the K subcarriers in one contiguous BW. In time domain, the IFFT modulates each data symbol onto a unique carrier frequency.

Explanations for A", B", and C" are the opposites of A, B, and C respectively

S/P: serial to parallel
P/S: parallel to serial
IFFT: inverse fast Fourier transform
FFT: fast fourrier transform
LO: local oscillator
D/A: digital to anolog converter
A/D: analog to digital converter

Figure 1.5 WiMAX transceiver.

$$e^{j2\pi f_c t} a(n) \tag{1.2}$$

where f_c is the carrier frequency.

Hence, the FFT algorithm is key in realizing the OFDM signal. Please refer to Figure 1.6.

1.9 OFDM Challenges and Mitigations

The OFDM technique currently is used in high-speed DSL modems over copper, as well as in IEEE 802.11g and IEEE 802.11a.

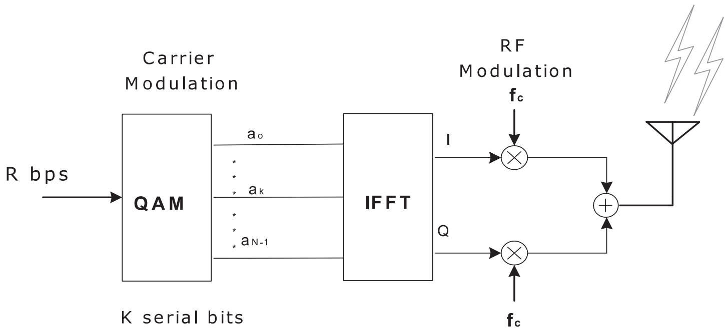
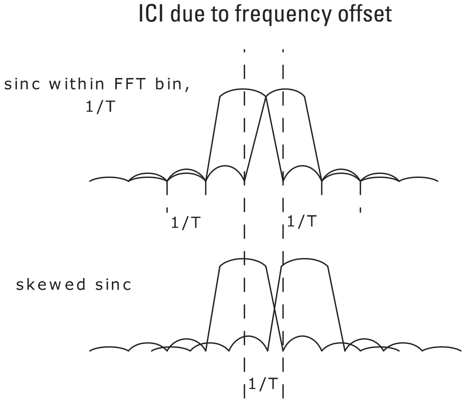


Figure 1.6 OFDM signal realization.

The attraction to OFDM is the need for an increased data rate. In an OFDM increase of BW, frequency-selective fading and the attendant signal distortion are inherent. However, to its strong credit, OFDM, through the use of low data rate subcarriers, mitigates ISI.

Remedying ISI, beyond low data rate subcarriers, using guard time helps avoid a big equalizer in the receiver. A guard time is added between two OFDM symbols, and when the FFT is taken in the receiver, there is no component before and after a symbol.

Frequency offset, between two subcarriers as shown in Figure 1.7, results in intercarrier interference (ICI). ICI is due to lack of time and frequency synchronization between transmitter and receiver. The effect of ICI is to cause a



In the first set of sinc functions, the max of the first is on the null of the next sinc function, while it is not the case in the second case.

Figure 1.7 FFT subcarrier skewing.

skewing of the FFT window in the receiver. ICI is caused by a loss of orthogonality between subcarriers.

One perhaps strong shortcoming of OFDM is a tendency to introduce amplitude variations in a transmitted signal when using QAM. Additionally, the very notion of the transmission of carriers in parallel encounters problems in implementation. However, the Fourier transform technique can, for the most part, help avoid this problem.

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2

OFDM Flavors, ISI, Doppler, Rayleigh, and Fading

2.1 OFDM Flavors

With a slight exception in FLASH-OFDM, various flavors of OFDM work similarly. In each case, the BW is split into many subcarriers (OFDM) and some number of subcarriers are grouped into subchannels (OFDMA and, to some extent, OFDM) and assigned to users.

OFDM, just as its parent FDM, divides the available frequency band into multiple subcarriers, the only difference being that, in case of the former, the tones are close, overlapping, and orthogonal. OFDMA goes further to group subcarriers into subchannels. S-OFDMA, adds the scalable component to OFDMA. That is, it scales the FFT size to the channel bandwidth, while keeping the subcarrier frequency spacing constant across different channel bandwidths. Channel bandwidth ranges from 1.25 to 20 MHz. Smaller FFT sizes are assigned to lower bandwidth channels, whereas larger FFT sizes are assigned to wider channels. These bandwidth ranges create flexibility to conform to operator, regulation and application requirements.

In the case of fixed WiMAX, as in OFDM, the constant number of subcarriers is 256. Even if we increase the bandwidth, the number of subcarriers remains at 256, but the bandwidth of each of the 256 subcarriers increases. Obviously, the act of increasing subcarrier bandwidth indiscriminately, while okay for Doppler spread, makes it difficult to mitigate ISI. This explains why fixed WiMAX is less fit for mobile applications.

Refer to Figure 2.1 showing an OFDMA subcarrier structure. A user close to the BS can be assigned more subchannels with a higher modulation scheme

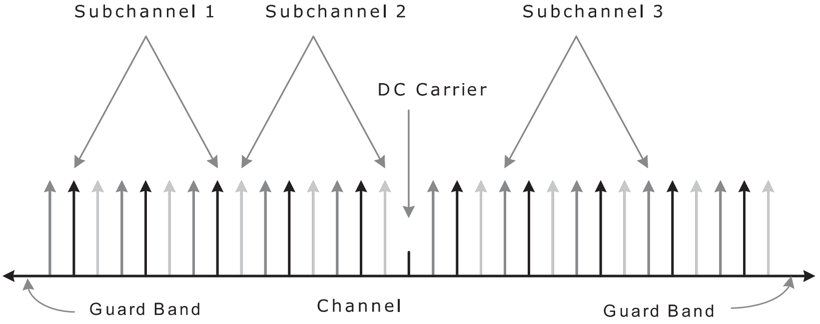


Figure 2.1 OFDMA symbol structure.

(for example, 64 QAM) using a high data rate. When a user moves further and further away from the BS, owing to degraded SNR, fewer subchannels are dynamically reassigned with lower and lower modulation schemes (as low as BPSK), thereby lowering the data throughput significantly.

The concept of scalability was introduced to the OFDMA mode (S-OFDMA) by the 802.16 Task Group e (TGe). A scalable physical layer enables standard-based solutions to deliver optimum performance in channel bandwidths ranging from 1.25 to 20 MHz, with fixed subcarrier spacing for both fixed and portable or mobile usage models, while keeping the product cost low. The architecture is based on a scalable subchannelization structure with variable fast Fourier transform (FFT) sizes, according to the channel bandwidth.

The frame structure under S-OFDMA improves system performance bandwidth. For scalability, tone spacing and symbol period are fixed, and only FFT size varies with system bandwidth. For system performance, the structure supports two new subchannels—band AMC subchannels for system throughput and safety subchannels for users in the cell boundary. In addition, dedicated control symbols are introduced for signaling efficiency. The first downlink symbol after the preamble is used for carrying system information, while the first three uplink symbols are assigned for channel quality report, ranging, and ACK/NACK of downlink data reception. Also, a new concept of bin and tile is introduced for carrier allocations of AMC subchannels and uplink diversity subchannels, respectively.

In S-OFDMA, which is applicable for a mobile environment, increasing the overall bandwidth will result in a commensurate increase in the number of subcarriers, while keeping subcarrier bandwidth constant at a compromised setting of 10.94 kHz. There is a good reason for this setting, as will be discussed later: it has to do with combating ISI and Doppler spread at the same time, a tradeoff solution.

Therefore, in S-OFDMA corresponding to different channel BW sizes, such as 1.25, 5, 10, and 20 MHz, there are 128, 512, 1,024, and 2,048 FFT sizes of subcarriers, respectively, but the subcarrier bandwidth remains 10.94 kHz in each case.

In most texts, the terms S-OFDMA and OFDMA are used interchangeably.

Targeting mobile applications also, FLASH-OFDM incorporates spread-spectrum technology to pseudorandomly (dependent on the channel condition) hop through the entire group of subcarriers to vary the subchannels assigned to users. With a frequency reuse of one, each BS uses the entire available spectrum. This process creates a strong frequency diversity and interference handling feature, not to mention better spectrum efficiency.

Last but not least, wireless broadband (WiBro) is a service name for mobile WiMAX in Korea and is based on IEEE 802.16e-2005. WiBro uses 9-MHz bandwidth in the TDD mode with a frame length of 5 ms. Refer to Table 2.1, which shows typical WiMAX OFDMA parameters.

2.2 Subchannelization

Subchannelization allows subcarriers to be disbursed to subscriber stations (SS) according to their data requirements and frequency responses.

Subcarriers are grouped into a subchannel in one of two rearrangement techniques: diversity permutation or contiguous permutation. Diversity permutation groups subcarriers into a subchannel pseudorandomly, thereby providing frequency diversity and intercell interference averaging.

Contiguous permutation, on the other hand, groups a subchannel by arranging subcarriers on a contiguous basis, without breaks or interruptions. In consequence, the adaptive modulation and coding (AMC) chooses the subchannel so arranged that has a better signal to interference + noise ratio (SINR) or a better frequency response.

Diversity permutation is for mobile applications, whereas contiguous permutation is for fixed or low-mobility applications.

Subchannelization allows all four variations (OFDM, FLASH-OFDM, S-OFDMA, and OFDMA) to split channels into subchannels, even into several thousand subchannels. Essentially, a user on an OFDMA network is assigned a number of subchannels across the band. A user close to the base station would normally be assigned a larger number of channels with a high modulation scheme, such as 64 QAM, to deliver the most data throughput to that user. As the user moves further away, the number of dynamically assigned subchannels becomes fewer and fewer. However, the power allotted to each channel is increased. The modulation scheme could gradually shift from 16 QAM to quadrature phase shift keying (QPSK) (four channels) and even binary phase shift keying (BPSK) (two channels) at longer ranges. The data throughput drops as

Table 2.1
WiMAX Parameters for a 10-MHz Bandwidth

Parameters	**Value**
System Channel Bandwidth (MHz)	10
Sampling Frequency (MHz)	11.2
FFT Size (N_{FFT})	1,024
Subcarrier Frequency Spacing (f)	10.94 kHz
Useful Symbol Time ($T_b = 1/f$)	91.4 ms
Guard Time ($T_g = T_b/8$)	11.4 ms
OFDMA Symbol Duration ($T_s = T_g + T_b$)	102.9 ms
Frame Duration	5 ms
Number of OFDMA Symbols	48
DL PUSC	
Null Subcarriers	184
Pilot Subcarriers	120
Data Subcarriers	720
Subchannels	30
UL PUSC	
Null Subcarriers	184
Pilot Subcarriers	280
Data Subcarriers	560
Subchannels	35

Source: [1].

the channel capacity and modulation change, but the link maintains its strength.

2.3 Scalable OFDMA, Doppler Shift, and ISI

S-OFDMA uses a compromised solution to mitigate Doppler shift and ISI to keep the subcarrier bandwidth constant when the overall bandwidth is varied. Additionally, S-OFDMA allows change (or scaling) of the bandwidth when the system grows.

2.3.1 Doppler Spread

For frequency ranges between 2 and 6 GHz, where the mobile WiMAX spectrum resides, Doppler spread is derived in the following way.

Assuming the worst-case scenario, let's take 6 GHz, as the higher the frequency, the more severe the Doppler shift. The Doppler shift is given by:

$$f_d = \frac{v_{vehicle}}{\lambda} \tag{2.1}$$

where $v_{vehicle}$ is speed of the vehicle and λ is the wavelength corresponding to the frequency at hand. Fist we compute λ as

$$\lambda = \frac{c}{f}$$

where c is the speed of light, given as 3×10^8 m/s and f is the frequency. Hence, for $f = 6$ GHz, $\lambda = 3 \times 10^8/6 \times 10^9 = 0.05$m.

For a vehicle speed of 122 km/hr (34 m/s), in the worst-case scenario, $f_d =$ $34/0.05 = 678$ Hz.

The coherence time, T_c, corresponding to f_d is given by:

$$T_c = \sqrt{\frac{9}{16.\pi.f_d^2}} \tag{2.2}$$

$T_c = \text{sqrt}(9/(16 \times \pi \times 678^2)) = 624\ \mu$s or $0.624\ \mu$s.

The 0.624 μs means that the symbols will stay undistorted as long as their duration is confined to 0.624 μs. Equivalently, 1/0.624 μs = 1.6 kHz can be thought of as data exchange and update rate of the channel. Doppler shifts also affect intercarrier interference (ICI) by way of exceeding the coherence bandwidth, as pointed out in Section 1.6.

2.4 Coherence Bandwidth

Coherence bandwidth, B_c, on the other hand, can be thought of as bandwidth in which multipath fading can be considered flat, and it is given by:

$$B_c = \frac{1}{5.\sigma_\tau} \tag{2.3}$$

where σ_τ is the maximum delay spread.

Coherence bandwidth corresponding to a worst-case delay spread, about 20 μs, is $1/5 \times 20\ \mu s = 10$ kHz. This means that the multipath fading within the 10-kHz subcarrier bandwidth is considered flat.

To mitigate ISI, a narrowband subcarrier (with a longer symbol time) is needed, while the mitigating Doppler shift requires a wideband subcarrier (with a shorter symbol time).

To get a sense of mathematical intuition consider the following:

Referring to the two formulas, (2.2) and (2.3), for coherence bandwidth and coherence time, we see that coherence bandwidth, in order to handle a conceivably larger delay spread, we require a narrower subcarrier bandwidth. From a coherence time perspective, a larger Doppler shift will require us to have a smaller coherence time (equivalent to a wider bandwidth) for the subcarrier. Since Doppler shift and coherence bandwidth are competing goals, in S-OFDMA they require a compromised solution.

2.5 ISI, Delay, and Fading

Let's consider other technologies, such as CDMA (IS-95). CDMA has brought numerous advantages to the cellular world, particularly voice application, in terms of mitigation of interference, increased capacity, and improved voice quality. Among these advantages are the use of a rake receiver to effectively take advantage of the adverse effect of multipaths, as well as the handling of soft and softer handoffs—the ability to connect with several sectors and BSs to carry the same call. The other virtue of CDMA is that, since various calls are separated in codes, it has the ability to extract conversation buried under noise and interferences. [CDMA first spreads its data bit into a pseudonoise (noiselike) wideband signal. That also explains why the transmit power is so low.]

The newer version of IS-95 is CDMA2000 with a 5-MHz bandwidth. The bit period is $T_b = 0.2\ \mu s$ $(1/5 \times 10^6)$, as opposed to IS-95 with a bit rate of 0.8 μs IS-95 would have a hard time handling a delay spread of 1 μs or more, if it were not for the rake receiver that assigns more hands for the delayed versions of the signal to add them constructively—one of the main fortes of the technology. On the other hand, resolving shorter delay spreads of 0.8 μs or less, as in dense urban or in-building applications, will be difficult for IS-95. That is where CDMA2000 comes in. In contrast to the older variant of the technology, it will be more appropriate for dense urban and in-building applications.

By comparison, in WiMAX, the low data rate used, implying a bigger bit period, will contain the delay spread and, hence, avoid ISI. On the other hand, the Doppler effect, which effectively causes the bandwidth of the channel to spill over, requires a lower bit rate, or a larger bandwidth in the frequency domain, to contain such a spill over. As these are competing goals, WiMAX/OFDMA simply has to implement a compromise solution to address ISI and Doppler shift problems.

2.5.1 Selective Frequency Fading Derivation

Refer to the Figure 2.2. In the figure, two paths are shown to the receiver. The first is a direct path, and the second is an indirect or reflected path. Path 1 takes t_1 time, while path 2 takes t_2, a longer time, to arrive at the receiver. The difference between t_2 and t_1 is τ: $t_2 - t_1 = \tau$.

The received signal, at the mobile receiver, from the two branches is $r_t(t)$ and is given as:

$$r_1(t) = \beta_1 \cdot u_1(t) + \beta_2 \cdot u_2(t) \tag{2.4}$$

where u_1 and β_1, u_2 and β_2 are the received signals and attenuations from path 1 and path 2, respectively.

Now replacing u_1 and u_2 by the same u yields the following:

$$r_t(t) = \beta_1 \cdot u(t) + \beta_2 \cdot u(t - \tau) \tag{2.5}$$

The attenuation in each case is different, but for the sake of simplicity, we will set $\beta_1 = \beta_2 = \beta$, and then $r_t(t)$ becomes $r_t(t) = \beta \cdot u(t) + \beta\ u(t - \tau)$.
Converting to frequency domain

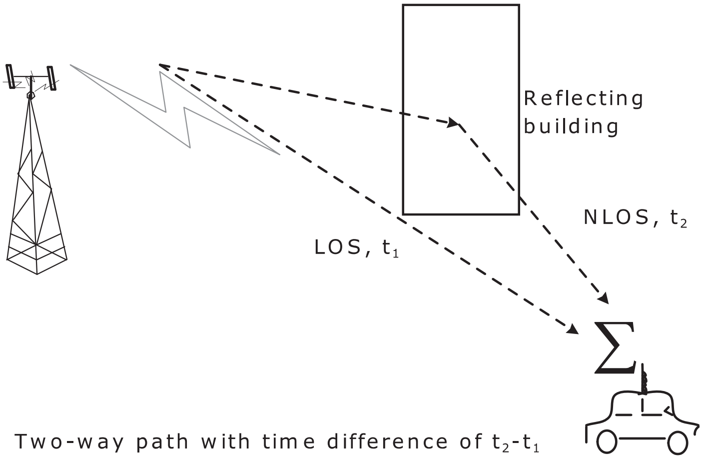


Figure 2.2 Multipaths.

$$
\begin{aligned}
R_t(t) &= \beta \cdot U(f) + \beta \cdot U(f) \cdot e^{-j2\pi f t} \\
R_t(f) &= \beta \cdot U(f)\left[1 + e^{-j2\pi f t}\right] \\
R(f) &= \beta \cdot U(f) \cdot H(f) \\
H(f) &= \left[1 + e^{-j2\pi f t}\right]
\end{aligned}
\tag{2.6}
$$

The polar form of the channel transfer function would be (as suggested by [2]):

$$H(f) = 2e^{-j2\pi f t} \cos\left(2\pi f (\tau/2)\right) \tag{2.7}$$

The magnitude part is:

$$\left|H(f)\right| = 2\cos\left(2\pi f (\tau/2)\right) \tag{2.8}$$

Plots of this magnitude as a function of frequency are shown in the following discussion related to frequency-selective fading versus flat fading.

2.5.2 Frequency-Selective Fading and Flat Fading

Frequency-selective fading is a case in which the channel spectrum is attenuated at various locations, depending on the delay spread of the channel. The more the delay, the more places the channel spectrum experiences attenuations or nulls.

Flat fading, on the other hand, is a phenomenon of a channel that causes the spectrum of a given transmitted signal to fade uniformly. This more or less uniform fading indicates an absence of multipaths to a large degree. Multipaths of short delays result in minimal frequency-selective fading. However, when the delays are pronounced, the channel magnitude suffers nulls at regular frequency intervals, effectively marking a point of entry into frequency-selective fading domain in earnest.

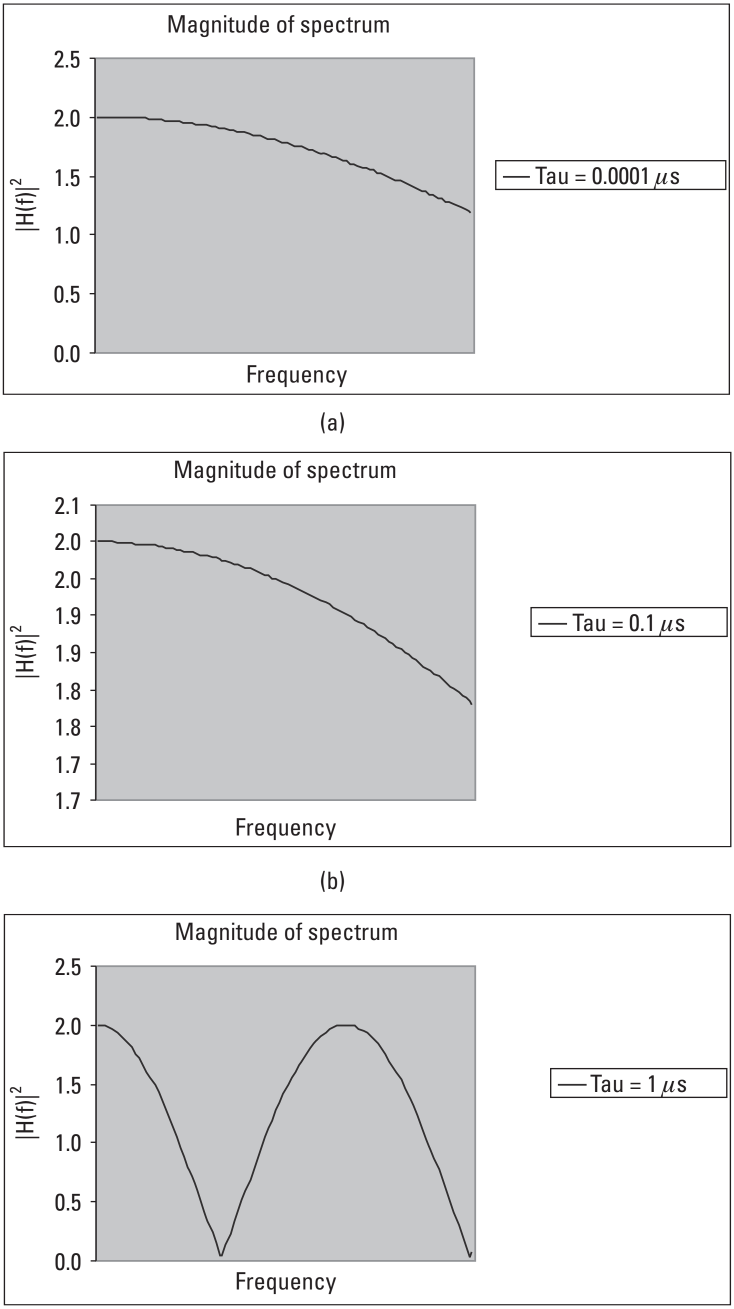
In Figure 2.3 are shown plots of frequency-selective fading as a function of τ(tau). τ is delay given in microseconds.

It can be observed that the more the delay, the more the channel (or the transfer function) experiences fading at more frequency locations.

As a cosine function, the magnitude of the transfer function of the channel is zero when the argument of the cosine function is at 90° or 270° or their respective multiples, or when the frequency is at $1/(2\tau)$, $3/(2\tau)$, and so on.

That is:

$$2 \cdot \pi \cdot f \cdot \tau/2 = 90° = \pi/2, \text{ implies } f = 1/2 \cdot \tau$$

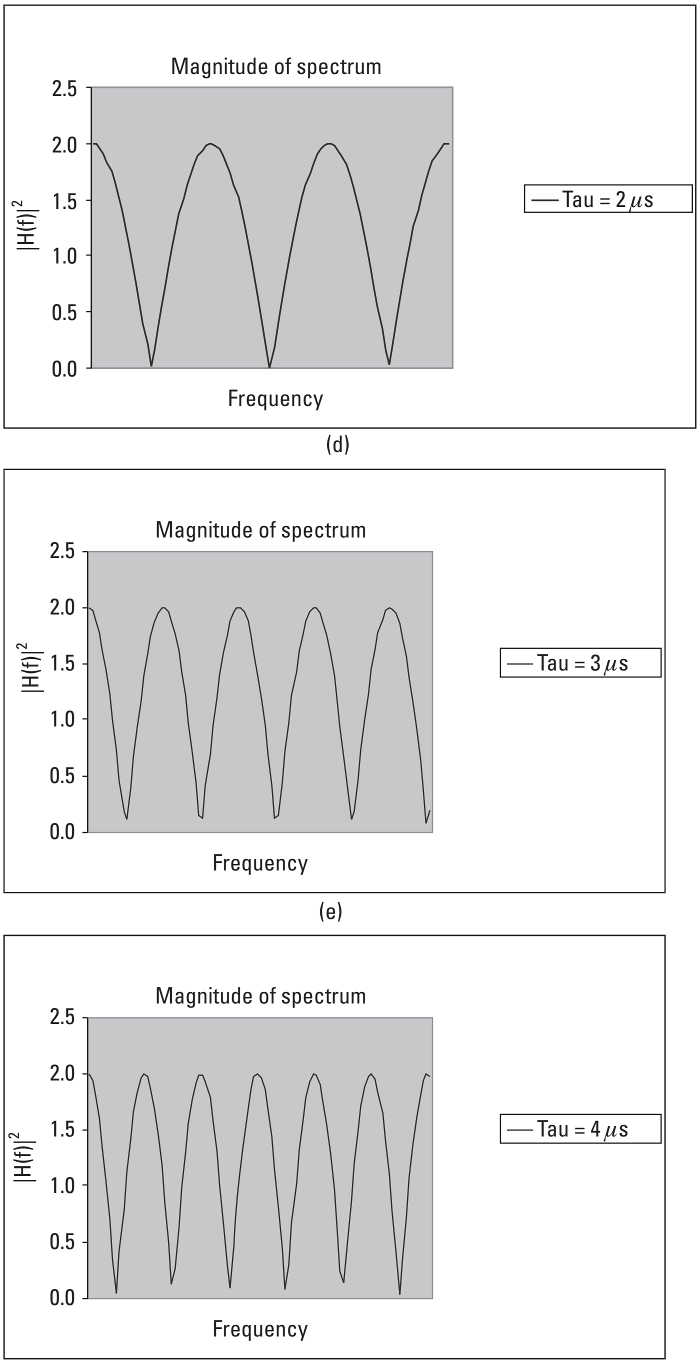


(a)

(b)

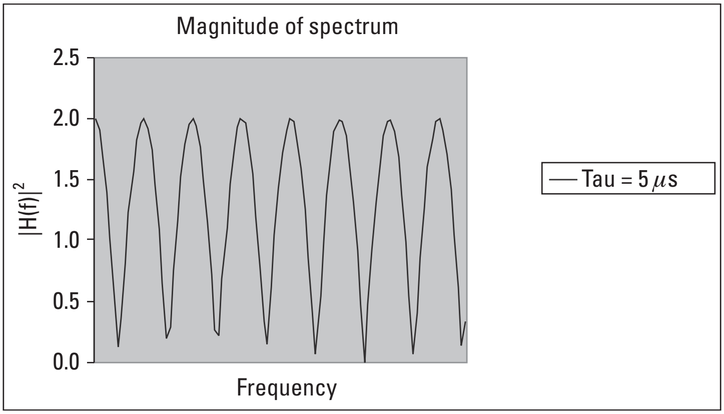
(c)

Figure 2.3 (a–h) Frequency-selective fading for various path delays.

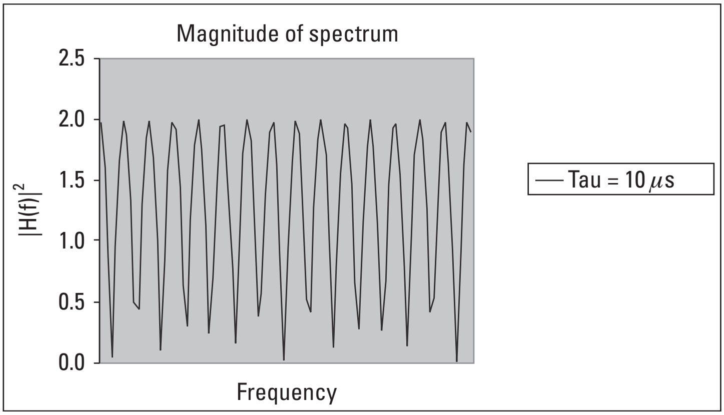


(f)

Figure 2.3 (continued)



(g)



(h)

Figure 2.3 (continued)

and

$$2 \cdot \pi \cdot f \cdot \tau/2 = 270° = 3\pi/2, \text{ implies } f = 3/2 \cdot \tau$$

The plots (a)–(h) in Figure 2.3 are effects of multipath (delay) in the frequency domain. The more the delay, the more the frequency-selective fading. Keep in mind the frequency scale on the *x*-axis, though not labeled, is the same in each plot.

2.5.3 Rayleigh Fading Simulator

Figure 2.4 shows a setup that closely simulates Rayleigh fading. Each signal branch is modulated by a Gaussian signal, which renders the signals random.

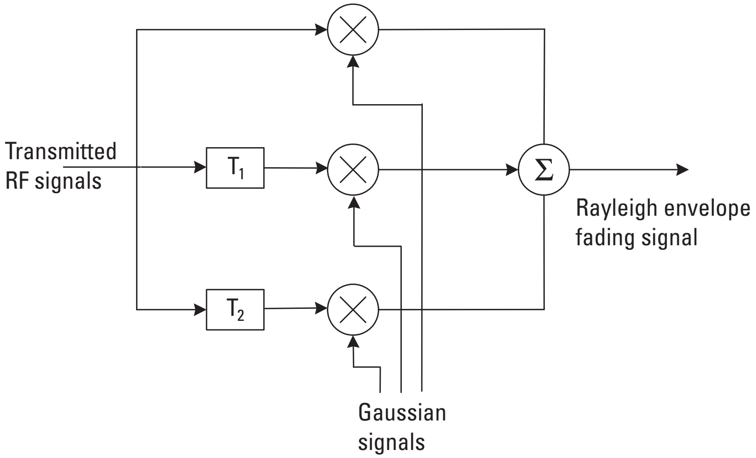


Figure 2.4 Rayleigh fading simulator with two delay components.

The Gaussian signal simulates the random attenuations and phases inherent in the channel, which can be thought of as the lognormal fading, along with the movement of the vehicle. The two branches additionally have delay components that simulate multipath signals taking alternate routes to arrive at the receiver. The addition of all three signals constitutes a Rayleigh envelope function closely simulating a Rayleigh fading signal.

2.5.4 Fade Rate and Doppler Effect—Slow Fading and Fast Fading Spread

Refer to the Figure 2.5 as we revisit the Doppler effect, but this time with a different slant, defining slow fading and fast fading:

f_d, the Doppler shift, is related to the velocity of vehicle and wavelength by:

$$f_d = \frac{V}{\lambda}$$

T_c, coherence time, is related inversely with f_d.

$T \approx \frac{1}{f_d}$ or $f_d \approx \frac{1}{Tc}$, hence, f_d or $f_d \approx \frac{1}{T_c}$ can be regarded as the fading rate of the channel. More accurately and specifically, T_c is related to f_d as $T_c = \sqrt{\frac{9}{16\pi f_d^2}}$, meaning the time duration thus obtained is a time duration over which the channel response has a strong correlation.

The rule of thumb or approximation of this is given by:

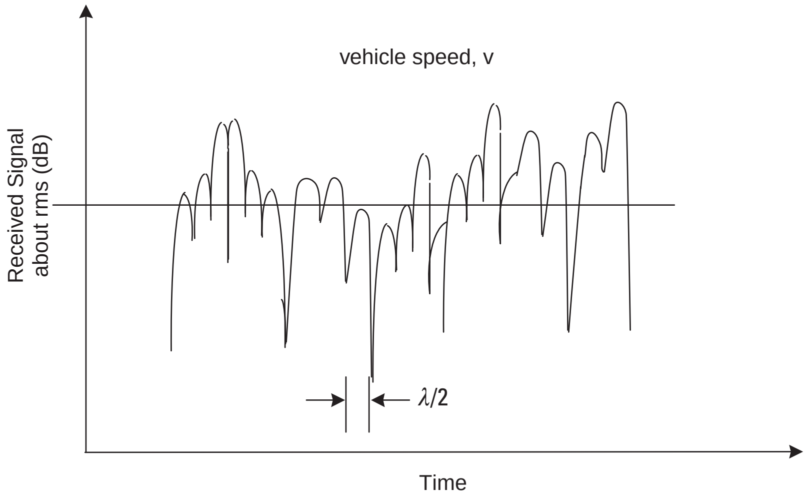


Figure 2.5 Rayleigh fading envelope.

$$T_c \approx \frac{0.5}{f_d}$$

As shown in the figure, the distance between the two nulls is $\lambda/2$, and time taken to cover $\lambda/2$ distance is:

$$\frac{\lambda/2}{V} \approx T_c = \frac{0.5}{f_d}$$

For example, for

$$f = 1{,}900 \text{ MHz (PCS frequency)}$$

V = 122 km/hr = 33.89 m/s
c, speed of light = 3×10^8 m/s
$\lambda = c/f = 0.158$m
Then

$$T_c = \frac{\lambda/2}{V} = 2.3 \text{ ms} = \frac{0.5}{f_d}$$

$$f_d = 214.5 \text{ Hz}$$

If, let's say, a voice channel has T_{symbol} (symbol rate) of 1,000 symbol/s, then we can conclude, since the fade rate, f_d, is considerably less than T_{symbol}, $f_d < T_{symbol}$, the channel experiences a slow fading effect.

If the opposite were true, that is, if $f_d > T_{symbol}$, then the channel is said to be experiencing fast fading. Fast-fading intervals can wipe out a series of consecutive symbols in a sequence, and the number of lost symbols drives the design of the interleaving and coding scheme.

References

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Part II
Technical Background

3

Modulation

3.1 Overview

What is modulation and why do we modulate a signal? Modulation simply implies translating by means of multiplying. Translating is necessary for the purposes of processing and facilitating sending to a particular position within a spectrum. This generally involves translating a baseband message signal to a bandpass signal at frequencies that are much higher. The baseband message signal is called the *modulating signal* and the bandpass signal is called the *modulated signal*. Modulation may be done by varying the amplitude, phase, or frequency of a high-frequency carrier in accordance with the amplitude of the message signal. *Demodulation* is the process of extracting the baseband message from the carrier so that it can be processed and interpreted by the intended receiver.

Digital/RF modulation, or bandpass modulation, is essential for the following major and compelling reasons to:

1. Keep antenna size small;
2. Work in harmony with other RF transmissions to contain interference and conserve spectrum;
3. Be able to bring a signal into a convenient frequency range for processing.

Imagine that we had a baseband signal we just processed within a WiMAX application in the 11-kHz range. Let's say we decided to transmit this low frequency data over the airwaves using an antenna. The antenna size we require is then:

$$\lambda = \frac{c}{f} = \frac{3\times 10^{8}\ \text{m/s}}{11\times 10^{3}\ \text{Hz}} = 24\ \text{km}$$

We need an antenna size of 27 km. If instead we translate the low frequency signal to one that is much higher in one of WiMAX bands, for example, 3.5 GHz, then we only need an antenna that spans:

$$\lambda = \frac{c}{f} = \frac{3\times 10^{8}\ \text{m/s}}{3.5\times 10^{9}\ \text{Hz}} = 8.6\ \text{cm}$$

These calculations show that higher frequency translation is critical, and it is made possible through the process of modulation.

Modulation is also essential for the purpose of regulatory and interference pollution control. If different entities requiring signal transmissions occupy an authorized spectrum with a given profile, this setup lends itself to better organization, better interference control and regulation of bandwidth, and cleaner airwaves. This is made possible by translating frequencies to small frequency slots to which stringent requirements are applied. Additionally, modulation offers the advantage of translating a frequency to a location where the spectrum is available. Last but not least, translation is essential and necessary when a signal can be brought to a location where design requirements, such as filtering and amplification, can be easily met. For example, in the receiver, the RF signal received is brought back to an intermediate frequency (IF) for processing [1].

Demodulation works in a similar fashion, either by reversing the process of modulation or, prior to that action, taking the signal into a processing frequency range such as IF, or both methods.

In contrast to digital modulation, analog modulation takes a cue from either amplitude variation (in case of AM modulation) or frequency variation (in case of FM modulation) of the message signal to vary the carrier wave. Refer to Figure 3.1.

3.2 Digital Modulation (Representing Digital Bits with Analog Signals)

In digital modulation, an analog carrier signal is varied by a digital bitstream. This process of allowing baseband digital data to modulate an analog carrier wave is called *digital modulation* (also called *bandpass modulation*). And the reverse, extracting the baseband message from the received modulated signal, is considered *digital demodulation*.

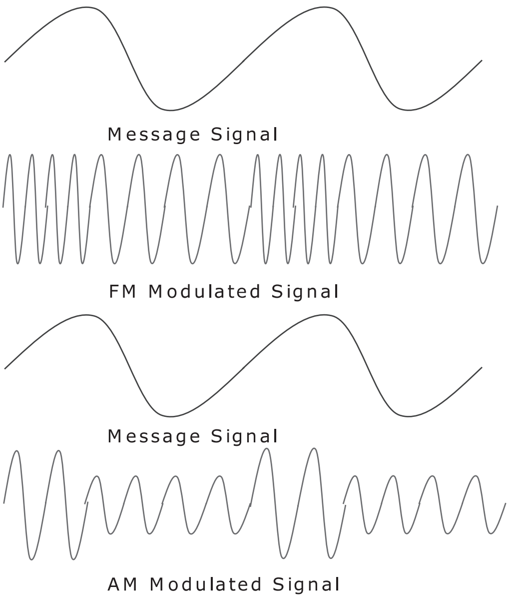


Figure 3.1 Analog modulation (FM and AM modulation).

Some of the clear advantages of digital modulation are better quality communications, higher information capacity, and compatability with digital data services, the main reasons for the 2009 digital TV migration.

There are three fundamental types of digital modulation (or shifting) schemes and one hybrid of the two schemes:

1. Amplitude shift keying (ASK);
2. Frequency shift keying (FSK);
3. Phase shift keying (PSK);
4. Quadrature amplitude modulation (QAM).

In all cases, it is important to emphasize that it is the digital message signal that causes the carrier signal to vary according to its value of either 0 or 1. The resultant waveform would look like that in Figure 3.2.

In the case of amplitude shifting, signal amplitude is varied to convey a difference between a bit of 0 or 1. The amplitude, for example, can be represented as twice as much when a 1 is being conveyed as when a 0 is being conveyed. Frequency modulation likewise represents 1 with a higher frequency than a 0. Phase shifting representation uses a signal with no shift to represent a 0, while a 1 is represented with a phase-shifted version of the same signal.

Each one of these shifting schemes has its own strengths and weaknesses.

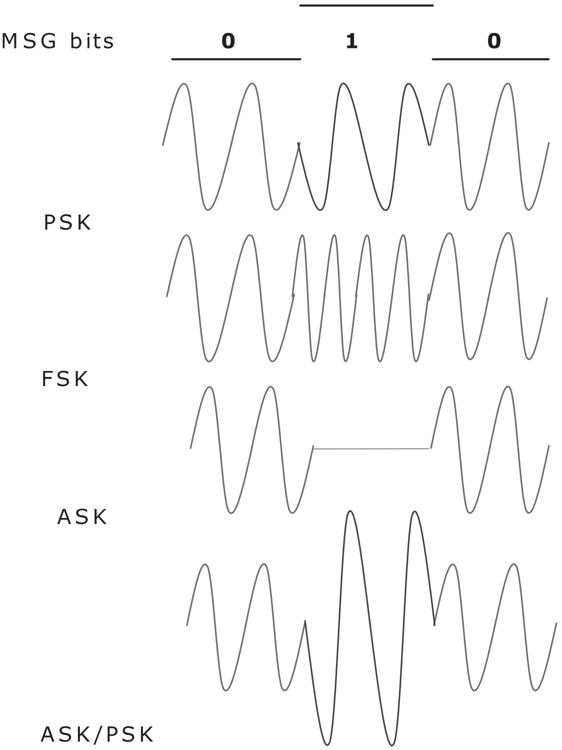


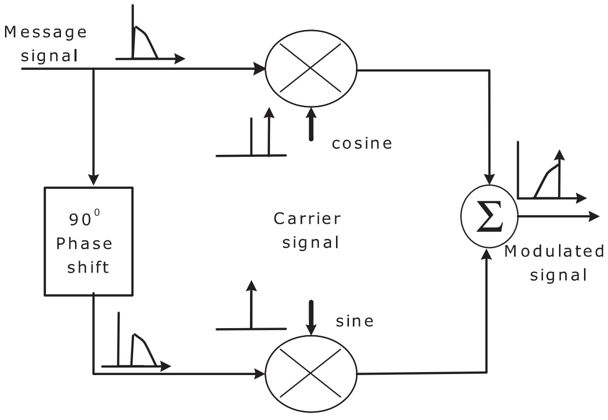
Figure 3.2 Digital modulation showing PSK, FSK, ASK, and ASK/PSK.

We judge their respective strengths and weaknesses according to their efficiency in power and bandwidth according to Shannon's capacity formulation.

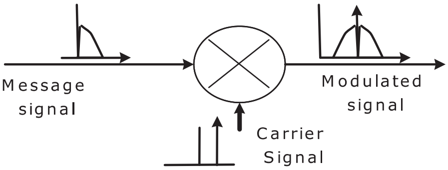
ASK is bandwidth efficient, as the signaling rate determines only the required bandwidth. However, it is liable to distortions in the presence of noise, and hence, is not power efficient. FSK, while it is power efficient, is not bandwidth efficient, as it depends on the signaling rate and the frequency chosen to represent the binary digital data. For example, if 1 is represented by a high-frequency signal, then that can lower the bandwidth efficiency significantly. If the signals as a whole have high frequencies they can also lower the bandwidth efficiency, but not any degree lower than the ASK scheme.

PSK is both power and bandwidth efficient. It is bandwidth efficient because it is dependent on only the signaling rates and no more. On account of using only phase changes, it doesn't succumb to noise distortion, and hence it is power efficient as well.

ASK/PSK, as shown in Figure 3.3, allows further bandwidth efficiency by combining the ASK and PSK schemes. QAMs are particular applications of both ASK and PSK.



Single sided modulator
(a)



Double sided modulator
(b)

Figure 3.3 (a) Single- and (b) double-sided phase shift keying concept.

Referring to Figure 3.3(b), we can see from the trigonometric identity in Appendix D, $\cos x \cos y = \frac{1}{2}\cos(x+y) + \frac{1}{2}\cos(x-y)$, that it generates a double-sided signal, which is comprised of two signals on either side of the carrier, at half the amplitude. The variable x represents the carrier and y represents the message.

On the other hand, referring to Figure 3.3(a), a message and a 90° shifted version multiply to cosine and sine oscillators, respectively.

In this case, two trigonometric identities apply:

$$\cos x \cos y = \frac{1}{2}\cos\left(x+y\right) + \frac{1}{2}\cos\left(x-y\right)$$

for the upper portion and

$$\sin x \sin y = -\frac{1}{2}\cos(x+y)+\frac{1}{2}\cos(x-y)$$

for the lower portion, whence they are added and result in 2(1/2 cos($x - y$) = cos($x - y$). As shown in Figure 3.3(a), this results in the lower single-sided message.

The following is the tradeoff between bandwidth efficiency and required power levels (power efficiency). According to Shannon's capacity formula relating capacity, *BW*, and power: $C = B\log_2\left[1+\frac{S}{N}\right]$, we see that capacity is related to bandwidth, *B*, linearly, while power, S/N, is related to capacity but not exactly linearly. If we manipulate Shannon's formula to represent bandwidth efficiency, C/B, then:

$$\frac{C}{B} = \log_2\left[1+\frac{S}{N}\right]$$

From this formula, we can see that there is a direct relationship between bandwidth efficiency and power efficiency. However, in the practice of designing a communication system, there is a tradeoff (or inverse) relationship, rather than a direct relationship. This inverse relationship is through the coding in two levels:

1. Incorporating error control coding to a message adds redundancy and overhead to the bulk of the message, and this consequently increases the bandwidth occupancy and reduces the bandwidth efficiency. At the same time, it reduces the required power level for a given bit error (now that we have introduced error control coding). Hence, this is the tradeoff.
2. On the other hand, higher level modulation schemes (*M*-ary keying) decrease the bandwidth occupancy but increase the required received power; hence, there is tradeoff between power and bandwidth efficiency. Refer to Table 3.1. *M* represents the number of possible signals in *M*-ary keying. For example, in *M*-ary = 8, the 8-PSK system requires a bandwidth of $\log_2 8 = 3$ times smaller than a BPSK system but a higher BER (requiring higher power).

3.3 Encoding and Decoding Phase Information

Suppose we have a message signal given by:

Table 3.1
Modulation Theoretical Bandwidth Efficiency

Format	Efficiency Limits (bits/s/Hz)
BPSK	1
QPSK	2
8 PSK	3
16 QAM	4
32 QAM	5
64 QAM	6
256 QAM	8

$$x(t) = A\sin(\omega t + \varphi)$$

We wish to encode the following data bits in the phase:

$$\varphi$$

Assuming a four-level, or QPSK modulation scheme, refer to Table 3.2.

To retrieve phase information, we can write:

$x(t) = \sin(\omega t +$  $)$ for 0° phase

$x(t) = \sin(\omega t +$  $)$ for 90° phase

$x(t) = \sin(\omega t +$  $)$ for 180° phase

Table 3.2
Bit Symbol and Phase Mapping

Phase Angle	Bits
$\varphi = 0°$	00
$\varphi = 90°$	01
$\varphi = 180°$	10
$\varphi = 270°$	11

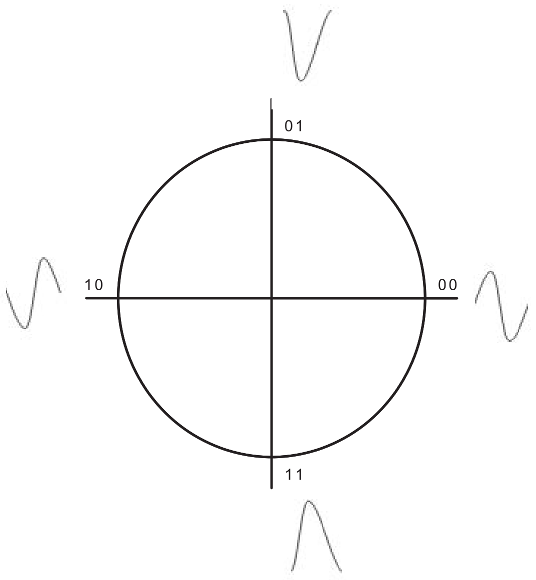


Figure 3.4 QPSK constellation.

$$x(t) = \sin(\omega t + \frown) \text{ for } 270° \text{ phase}$$

Now, at the receiver, let the local oscillator be of the form:

$$y(t) = \sin(\omega t)$$

Upon multiplying the incoming signal $x(t)$ with $y(x)$, we have:

$$\begin{aligned} y(t) \cdot x(t) &= \sin(\omega t) \cdot \sin(\omega t + \varphi) \\ &= \frac{1}{2}\cos(2\omega t + \varphi) + \frac{1}{2}\cos(\varphi) \end{aligned}$$

Right away we observe that the phase is decoupled from the rest and that an additional high-frequency component, namely, $2\omega t$, is introduced. The high frequency component can be easily filtered out and the phase easily processed.

There is only one problem: Can a phase shift of 90° or –90° be easily distinguished? No! Either 90° or –90° yield the same result, namely, 0 (zero). That is, 1/2cos(90°) = 1/2cos(–90°) = 0. Put another way, a phase shift with a cosine of 0 is either 90° or –90°. Hence, we need additional information in order to accurately decode phase shifts present in all quadrants.

Supposing we prepare an additional oscillator, a phase-shifted version of the original

$$\sin(\omega t), \text{ that is } \cos(\omega t)$$

Now we have the following equation in addition to the previous one:

$$y(t)\cdot x(t) = \sin(\omega t)\cdot \sin(\omega t + \varphi)$$

From trig identity:

$$y_2(t)\cdot x(t) = \cos(\omega t)\cdot \sin(\omega t + \varphi)$$
$$\cos(\omega t)\cdot \sin(\omega t + \varphi) = \frac{1}{2}\sin(\omega t + \varphi) + \frac{1}{2}\sin(\varphi)$$
$$\cos(\omega t)\cdot \sin(\omega t + \varphi) = \frac{1}{2}\sin(\omega t + \varphi) + \frac{1}{2}\sin(\varphi)$$
$$x(t)\cdot y_2(t) = \frac{1}{2}\sin(2\omega t + \varphi) + \frac{1}{2}\sin(\varphi)$$

Combining the two equations yields the following:

$$x(t)\cdot y(t) + x(t)\cdot y_2(t) = \sin(\omega t)\cdot \sin(\omega t + \varphi) + \cos(\omega t)\cdot \sin(\omega t + \varphi)$$

Then, from identity, we have:

$$x(t)\cdot y(t) + x(t)\cdot y_2(t) = \frac{1}{2}\cos(2\omega t + \varphi) + \frac{1}{2}\cos(\varphi) + \frac{1}{2}\sin(2\omega t + \varphi) + \frac{1}{2}\sin(\varphi)$$

The high-frequency components can be filtered out, and the phase angle in any quadrant can be determined. This will determine the exact phase of modulation encoding prior to transmission, whereas upon demodulation, we have a decoupling of the phase that can be detected in terms of voltage levels.

Let’s take this one step further. From the last step, we see that a pair of high-frequency waveforms at half the amplitude or voltage of the original and two phase levels at half the amplitude levels are transmitted and received. Clearly, it takes a lot of power to transmit a pair of high-frequency components and then filter them out. However, that is the price to detect the phase.

Even more critical, if we intentionally branch the datastream, so that odd and even bits go to either the sine or the cosine oscillator, respectively, to be modulated separately then we have effectively conserved bandwidth. In the previous case, we let the same data stream (one branch shifted by 90°) go to the sine and cosine branches of the oscillators. Now we are using different bits of the same data stream—even and odd bits, respectively. Not only does this conserve bandwidth, but also the data branches are orthogonal. This bears a lot of similarity to WiMAX carrier modulation, in which the tones are very close to each other, but they are also orthogonal. Refer to the QPSK in Figure 3.5, showing the modulator, where the message bits enter a separator (demultiplexer). The odd bits are branched down to the sinωt oscillator, and the even bits are directed to the cosωt oscillator on top. Upon modulation, the respective oscillators and the two added branches yield a tightly close, yet orthogonal signal, thereby conserving bandwidth.

3.4 Quadrature Amplitude Modulation

QAM is a method of combining two amplitude-modulated (AM) signals into a single channel, thereby doubling the effective bandwidth. In a QAM signal, there are two carriers, each having the same frequency but differing in phase by 90° (one-quarter of a cycle, from which the term *quadrature* arises). One signal is called the I signal, and the other is called the Q signal. Mathematically, one of the signals can be represented by a sine wave and the other by a cosine wave. The

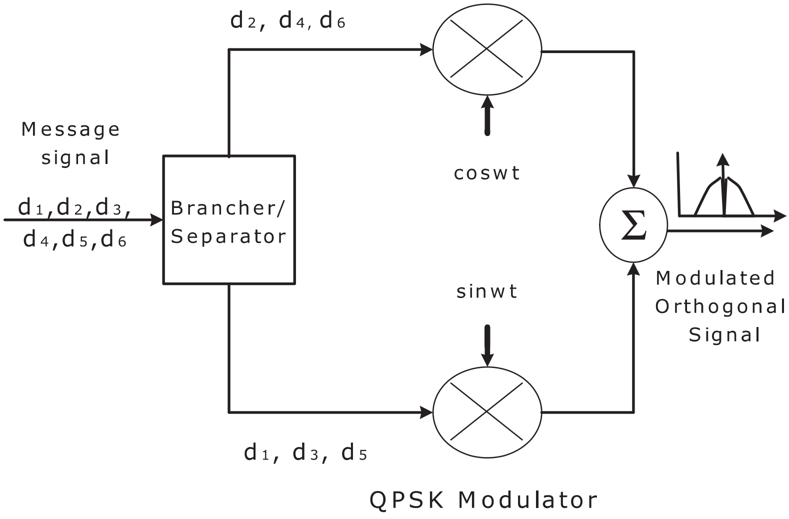


Figure 3.5 QPSK modulator.

two modulated carriers are combined at the source for transmission. At the destination, the carriers are separated. The data is extracted from each, and then the data is combined into the original modulating information.

QAM is a simple and yet effective high data rate modulation scheme that combines changes in both phase and amplitude to encode data bits. Some common QAM modulations are 16 QAM, 32 QAM, 64 QAM, and even 256 QAM. Refer to Figure 3.6, which shows 16 QAM. The further the distance is from the origin, the greater the amplitude, and the more the angle from the reference horizontal line, the greater the phase shift. The horizontal line represents 0° phase, while the vertical line represents a 90° phase shift. In the figure shown, there are 16 points representing various phase shifts and amplitude sizes. Specifically, amplitudes and phase shifts are illustrated corresponding to 0° and 45°, respectively.

Each state is defined by a specific amplitude and phase. Each time the number of states per symbol is increased, the total data and bandwidth increase. However, the higher the number of states per symbol, the more susceptible it is to Gaussian noise corruption and mistaken detection. That is because the individual signal representations are more tightly packed in the constellation.

For example, 64 QAM is more tightly packed than a 16 QAM constellation. While such packing will result in a higher data rate, since now the

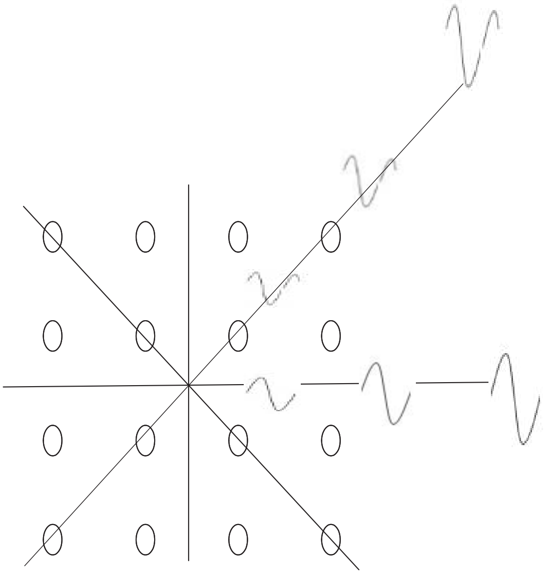


Figure 3.6 16-QAM signal space showing phase and amplitude variations—illustrated, 0° and 45° phases and amplitude, increase according to distance from origin.

individual signals are closer together, background noise can readily blur the distinction between any two adjacent signals. Encoding of QAM is done according to polar coordinates by angle and distance.

3.5 Multicarrier Modulation Transmissions and OFDM

The obvious benefit of multicarrier modulation is a high data rate. As discussed in more detail elsewhere in this book, especially in Chapter 2, selective frequency fading, as opposed to flat fading, is more pronounced when the delay between the primary signal and its reflected version is large. Equivalently, when the signal bandwidth is larger than the coherence bandwidth. Also directly related, ISI is caused when the symbol period of a baseband signal is less than the maximum of the delay spread of the channel.

In order to combat ISI, the multicarrier modulation scheme divides the high-speed data rate into multiple low data rate streams. This effectively makes the symbol rate larger than the maximum delay spread for each substream. (See the following example.) Refer to Figure 3.7, which shows multicarrier transmission.

An alternative and yet effective implementation of multicarrier transmission is using IFFT, and it is called OFDM. An efficient means of implementing spectrum-efficient and orthogonal subcarriers, OFDM uses only a single radio. Figure 3.8 shows a simple OFDM modulation and implementation technique. The receiver uses FFT to reverse the operation.

Example of an OFDM Application [2]:

Given bandwidth BW, 1 MHz; filter roll of factor 0.25; delay spread = 20 μs; design a number of carrier N that guarantees an ISI-free system using 64 QAM.

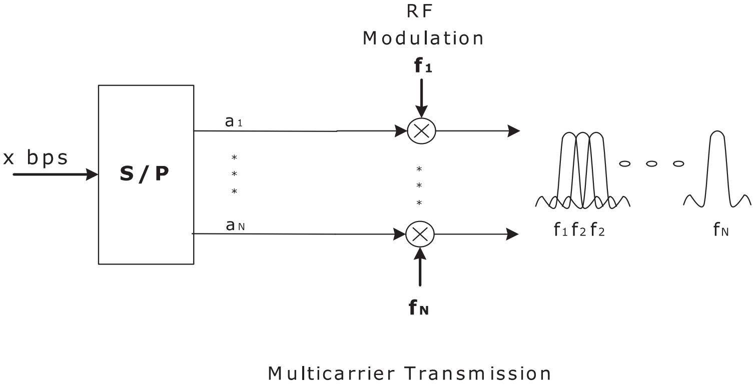


Figure 3.7 Multicarrier modulated transmission.

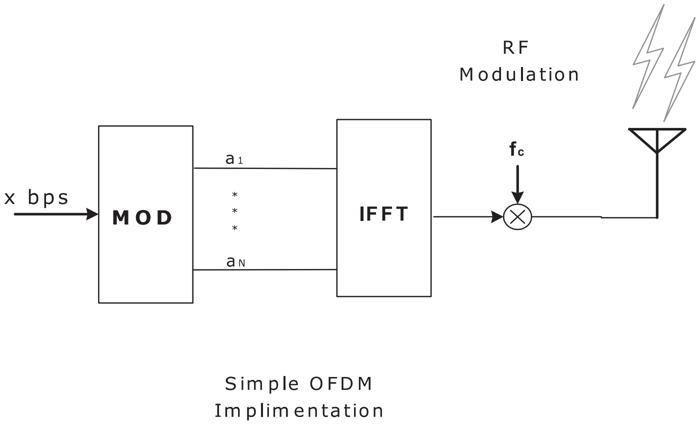


Figure 3.8 Simple OFDM modulation technique.

$$\text{BaudRate} = \text{SymbolRate} = \frac{BW}{1+\gamma} \tag{3.1}$$

where γ is the filter roll of factor and BW is the transmit bandwidth.

$$\text{Raw bit rate} = (\text{bits/baud})\cdot\text{baud} \tag{3.2}$$

$$\text{Data bit rate} = \text{FER code rate}\cdot\text{Raw bit rate} \tag{3.3}$$

Therefore, from (3.1), baud = 2 MHz/1.25 = 1.6 Mbaud

From (3.2) and bits/baud = 6 (for 64 QAM), raw bit rate = 6·1.6 Mbaud = 9.6 Mbps.

From (3.3) assuming code rate = 1/2, data rate = 1/2·9.6 Mbps = 4.8 Mbps.

To accommodate a 20-μs delay spread with 10 times the maximum delay spread, $10 \cdot 20\,\mu\text{s} = 200\,\mu\text{s}$.

Then $T_s = \dfrac{1}{\Delta f} = \dfrac{1}{\frac{BW}{N}} \geq 200$

where $N = 200$, implying a carrier spacing of 5 kHz corresponding to a 200-μs symbol spacing that ensures ISI-free communication.

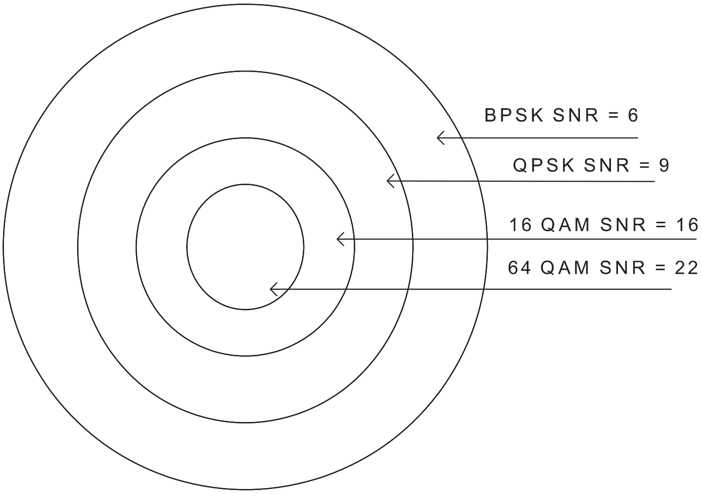


Figure 3.9 Cell radii relative to AMC.

3.6 Adaptive Modulation and Coding (AMC)

The WiMAX system supports adaptive modulation [3]to regulate the signal modulation scheme (SMC) depending on the signal-to-noise ratio (SNR) state of the radio link. When the radio link is high in quality, a peak modulation scheme is used, thereby improving capacity. Refer to Figure 3.9. During low SNR or fade conditions, the system switches to a lower modulation scheme, maintaining link stability and connection quality. In contrast to using a modulation scheme for the worst possible link conditions in other systems, WiMAX optimizes the modulation scheme to reflect the fading conditions that effectively enhance the extent and range in which a higher modulation can be used.

However, despite the adaptive modulation mechanism to account for variations of SNR, AMC would not be able to correct such anomalies as a lack of frame synchronization and a varying UL/DL ratio from BS to BS.

References

[1] Sklar, B., *Digital Communications: Fundamentals and Applications*, Englewood Cliffs, NJ: Prentice-Hall, 1988.

[2] Rappaport, T. S., *Wireless Communications: Principles and Practice*, Upper Saddle River, NJ: Prentice-Hall, 1996.

[3] Dietze, K., et al., "WiMAX Uplink and Downlink Design Consideration," EDX Wireless Technology, white paper, 2008.

4

RF Propagations, Measurements, and Models

4.1 RF Propagation

In this section, we shall briefly discuss the characteristics of radio frequency propagation, define cell site ranges, confidence level, and intervals of coverages.

4.1.1 Cell Site Definitions

The very fundamental idea of cellular technology, at least when it was first conceptualized, pivots around the idea that by reducing the transmit power (and as a result, reducing the service area) the reuse of a frequency *again and again* is possible and helps increase capacity and conserve the premium spectrum. However, different applications require different transmit power and sizes of cell sites.

Definition of cell sizes:

1. Macro cell;
2. Micro/pico cell;
3. Femto cell.

Macro cells are for outdoor and in-vehicle applications. The cell site locations are usually on towers, buildings, or water tanks. The coverage radii range from 1–30 km. *Micro cells* (though smaller in radius, sometimes the term is interchanged with "pico cell") are for coverage of localized regions. This might include airport terminals, train stations, office complexes, or college campuses.

Micro cells (and pico cells) can be used as a means of signal retransmission of macro cells into a confined area, such as a conference hall or a particular floor in a building. A good example of a micro cell is a WiFi hotspot in a Starbucks store, where the clients share the signal. Micro cell radii are between 200 and 2,000m, while pico cells are between 4m and 200m. A *femto cell* is the smallest in radius of coverage and is comparable or analogous to a WiFi home router or antenna range inside a residence.

Using smaller cell sites can increase capacity as well as transmit power while increasing switching activity on account of frequent handoffs. Another downside to reduction of cell radii is the increased complication of the E911 application used to locate subscribers in an event of emergency.

4.1.2 Free Space Path Loss

Free space path loss, as the name implies, is an idealized model that takes into account path loss in free space. In most instances, as discussed rather adequately elsewhere in this text, free space path loss connotes a loss of about 20 dB per decade (20-dB loss for every 10× distance traversed), all other factors remaining constant. In Figure 4.1, the sloping straight line (labeled Path Loss) approximately represents a free space path loss, while the curve labeled Shadow Fading, represents the superposition of obstruction loss, modeled as a lognormal function, on top of distance-related loss. The curve labeled Rayleigh Fading represents the multipath effect, yet another superposition on top of the other two curves just mentioned. It is nearly always a good idea to start with the idealized

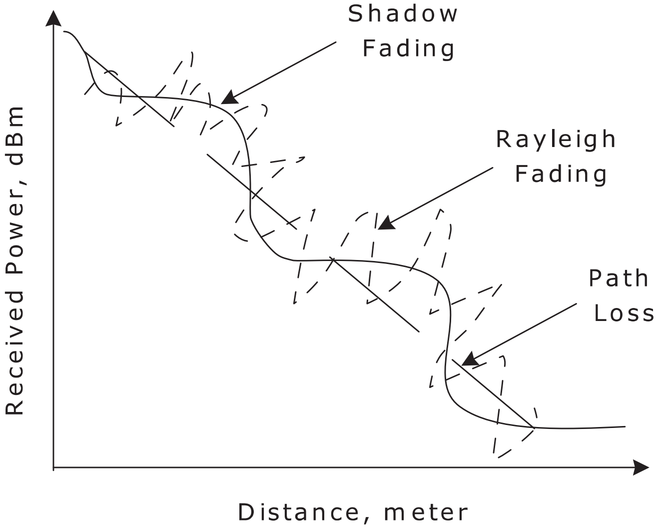


Figure 4.1 Received power versus distance.

version of loss computation, namely, free space path loss, and then account for the other types. It should be emphasized that free space path loss starts its own linear region at a reference distance, d_o, below which (closer to the antenna) this idealized model, on account of the nonlinear nature of antenna propagation, cannot apply.

4.1.3 Confidence Level

Generally it is in the interest of any service provider or designer to ascertain coverage confidence levels—for contractual reasons, for example. Referring to Figure 4.1, for a dataset collected at a specific radius, d_i, the mean power level is given by [1]:

$$P_{d_i} = P_0 - \gamma \log \frac{d_i}{d_o} \tag{4.1}$$

where P_0 is the power level at a referenced distance, d_o, and γ is the path loss exponent. If, for this given mean power and a given standard deviation, σ, 8 dB typically, and radius represented as a random variable, x, then we can calculate the confidence level relative to this random variable. Hence, using the standardized probability function, the confidence level can be computed as:

$$P(x \geq l) = P\left(\frac{x - P_{d_1}}{\sigma} \geq m\right) = confidence\_level \tag{4.2}$$

$$P(x \geq l) = \int^{\infty} \frac{1}{\sqrt{2\pi}\sigma} e^{-\frac{\left(x - P_{d_1}\right)^2}{2\sigma^2}} dx = confidence\_level \tag{4.3}$$

If the l (threshold) is increased for the same mean received power P_{d_1}, then the confidence will go down, and vice versa.

For example, if P_{d_1} were –80 dBm and a desired confidence level were 30%, what should the signal level be equal to or greater than ($x \geq$ l)? We can pose the question differently as well. Given a mean signal power, for a signal level above the mean x and above, by how much will my confidence level decrease?

Looking at a standard table, or $Q(z)$ function table, corresponding to 30% probability, $z = 0.5$

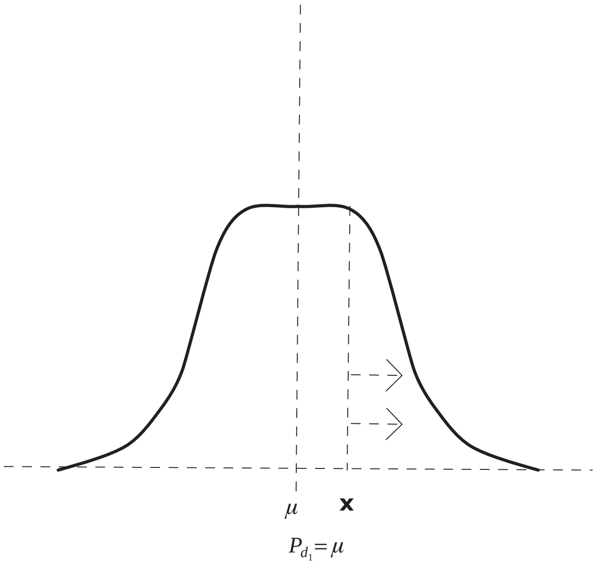


Figure 4.2 A normal Gaussian curve showing confidence level.

$$
\begin{aligned}
&P\left(\frac{x - P_{d_1}}{\sigma} \geq 0.5\right) = 30\% \\
&x - P_{d_1} \geq 0.5\sigma \\
&x \geq 0.5\sigma + P_{d_1} \\
&x \geq 0.5(8) + (-80\text{ dBm}) \\
&x \geq -76\text{ dBm}
\end{aligned}
$$

For a confidence level of 50%:

$$
\begin{aligned}
&P\left(\frac{x - P_{d_1}}{\sigma} \geq 0\right) = 50\% \\
&x - P_{d_1} \geq (0)\sigma \\
&x \geq P_{d_1} \\
&x \geq (-80)\text{ dBm} \\
&x \geq -80\text{ dBm}
\end{aligned}
$$

So that confirms that the mean value we started out with, –80 dBm, corresponds to a confidence level of 50%. Increasing the signal levels, as in the previous example, resulted in a lowered confidence level, namely, 30%.

Now, if we desire higher confidence, we should expect a lower signal level. For an 80% confidence level:

$$P\left(\frac{x - P_{d_1}}{\sigma} \geq -0.85\right) = 80\%$$
$$x - P_{d_1} \geq (-0.85)\sigma$$
$$x \geq (-0.85)8 - (80\ \text{dBm})$$
$$x \geq -86.8\ \text{dBm}$$

Incidentally, at a design level, working with a link budget, adding fade margin padding will always increase the confidence level. So, a zero margin, as we saw earlier, results in a confidence level of 50%, while additional margins will increase the confidence level proportionately.

4.1.4 Confidence Interval

If received power data is collected, via drive testing, and the pieces of data pertaining to a particular radius, d_1, are pegged, the mean for this data set is

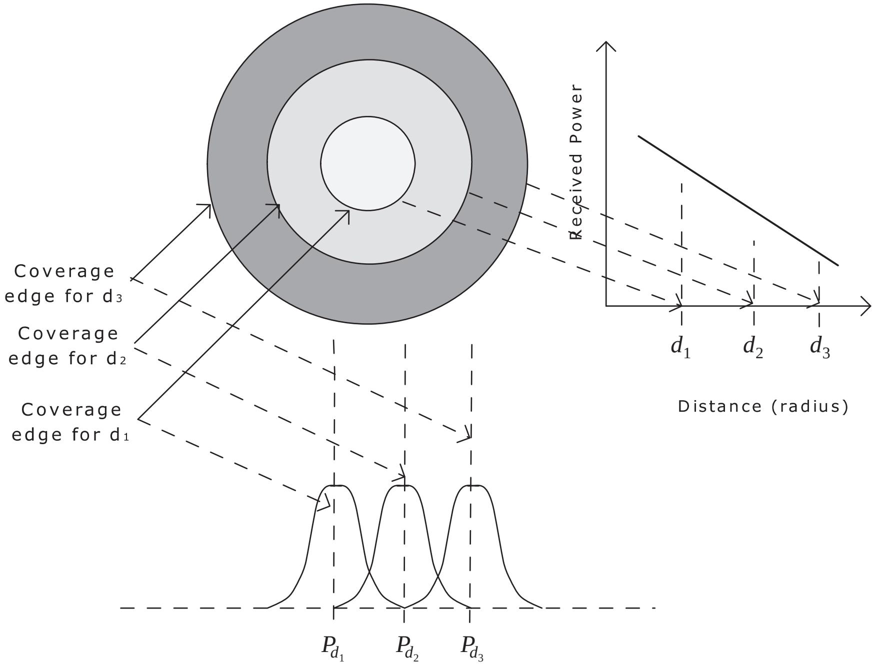


Figure 4.3 Confidence level for a given mean power relative to a given distance.

computed and named, P_{d_1}. It is clear the data varies according to the standard deviation σ or a fraction thereof. In fact it varies in either direction, plus or minus, by $n\sigma$ to $-n\sigma$, ignoring the fractional variations that lie between them.

The confidence interval deals with exactly such a variation. Once the confidence level is computed with respect to the standard deviation:

$$P(x \geq l) = P\left(\frac{x - P_{d_1}}{\sigma} \geq m\right) = \mathit{confidence\_level} \tag{4.4}$$

where l is the random variable above which the probability will result in the desired confidence level (given in a percentage), and m is the standardized normal random variable. The confidence interval then goes one step ahead in taking into account the variation in either direction of the mean receiver power level, P_{d_1}.

The mean is the local mean that the transient signal, or more particularly, the Rayleigh fading envelope superimposes. For a standstill mobile application, a more or less constant signal is observed. When the vehicle moves, multipath signals, that constitute the Rayleigh fading envelope, are observed. As the vehicle moves faster, the rate of fading becomes more pronounced and agitated.

$$P\left(P_{d_1} - \sigma \leq x \leq P_{d_1} + \sigma\right) = \int_{P_{d_1}-\sigma}^{P_{d_1}+\sigma} \frac{1}{\sqrt{2\pi\sigma}} e^{-\frac{(x-P_{d_1})^2}{2\sigma^2}} dx = 68\%$$

The previous equation states how often the signal varies around its mean from $P_{d_1} - \sigma$ to $P_{d_1} + \sigma$. Refer to Figure 4.4, which shows the confidence interval.

4.2 Measurements and Drive Test Considerations

Drive test processes usually collect, store, and displace a continuous received power corresponding to a location identified with the GPS receiver as a vehicle follows a predetermined driving route.

Drive tests and measurements are necessary at various phases of network design. True, prediction tools will help avoid frequent and expensive drive tests in exchange for a somewhat less accurate but quick reflection of the reality.

We absolutely need drive test data for the following reasons:

1. To tweak propagation models so that they are equipped with the correction factors;

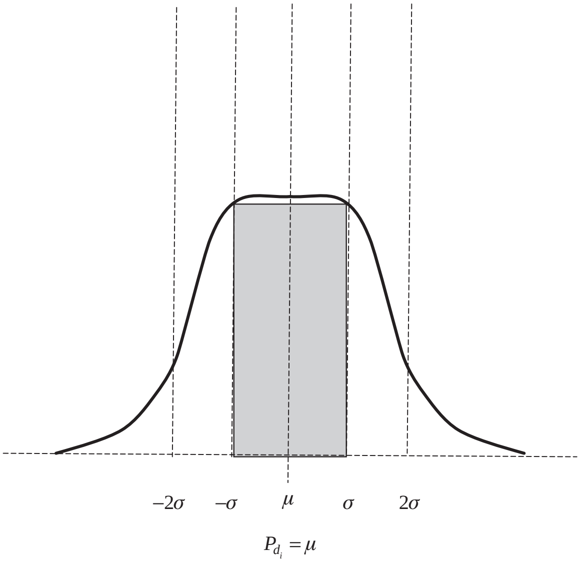


Figure 4.4 Normal curve showing confidence interval relative to mean power.

2. To measure actual performance:
    (a) Optimization;
    (b) Benchmarking;
    (c) Interference characterization.

Performing a drive test, in order to be successful and avoid repetitive visits, requires proper planning and assembling of the necessary equipment. For the most part, the following are needed:

- A GPS receiver;
- A mobile receiver or handset similar to the one used in the system;
- An antenna and cable connecting to the roof of the vehicle;
- A laptop computer to receive, format, display, and store signal (received power) and GPS location (latitude and longitude) information;
- A determination of the vehicle's speed;
- A planned drive route;
- A methodology of processing the data collected.

Processing of the drive test can be done with different software back in the office or lab after transferring the data from the laptop used for collecting it.

The processing software can take the collected received power versus location data and can, for example, plot it on a map, show mutual interference and handoff locations, and so on.

In this discussion, we look only at the confidence-level and confidence interval aspects of the measurement data. For example, a designer would want to know the service area reliability. In other words, for a given signal threshold, say, –85 dBm, how many data entries of –85 dBm or greater are recorded, compared to the total data entries recorded?

For a vehicle speed of 65 miles/hr = 65·1.6 km/hr = 65·1.6·1,000 = 104,000m/3,600s = 29 m/s.

Given a 2.5-GHz wavelength, $\lambda = 0.12$m, if, for example, as shown in Figure 4.5, every average sample *n* within $50 \cdot \lambda = 6$m, one average sample will be collected. The car traverses 29/6 samples or about 5 samples per second.

If, for example, we desire a 90% confidence level of our mean, then we can figure out the number of samples, *n*, from using the following relationship [2]:

$$P_r\left(p' - c < p < p' + c\right) = 0.90$$

where p' is a sample proportion or fraction of items of interest in a sample. This also can be viewed as a probability of success of y in n trials, ($p' = y/n$). The parameter c is very important. It is a half width of the confidence interval. Our estimate of the mean will hopefully contained within the confidence interval. It

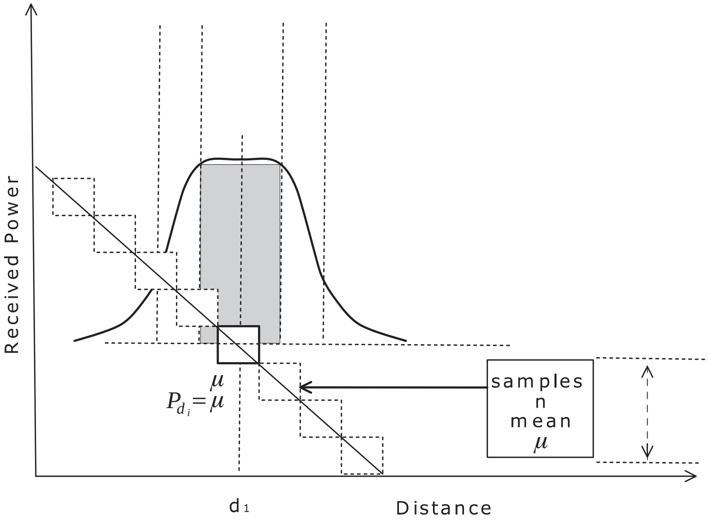


Figure 4.5 Samples and confidence interval for drive test.

is very certain that our estimation of the mean from a probability vantage point will never equal the estimation of the true mean. The parameter *c* gives us the insight. In order for our exercise to be a worthwhile effort, the true mean must be embraced. If we select our *n* we can ensure that it happens.

Hence, a 90% confidence interval is between the following two limits [2, 3]:

$$p' - 1.645\sqrt{\frac{p'(1-p')}{n}} \text{ to } p' + 1.645\sqrt{\frac{p'(1-p')}{n}}$$

where

$$c = 1.645\sqrt{\frac{p'(1-p')}{n}}$$

The interval value corresponding to the confidence level of 90% is 1.645, from a symmetric normal curve, such as $\Pr(-a < z < a)$. In our case, $a = 1.645$. $\sqrt{\frac{p'(1-p')}{n}}$ is the standard deviation, σ_p, of our estimate, p'. Hence, *c* is given as $c = a\sigma_{p.}$ Then solving for *n*,

$$n = (1.645)^2\,\frac{p'(1-p')}{c^2}$$

The only problem is that we don't know p'. Simplifying, $p'\ (1 - p')$ does not exceeding 0.25. That is, no matter the value of p', the product $p'\ (1 - p')$ cannot exceed 0.25. Then, with this worst-case scenario approximation, *n* numerically would be (assuming a reasonable value for *c* +/– 0.022 dB): $n = (1.645)^2 \cdot 0.25/0.022 \cdot 0.022 = 1{,}398$ samples. So, we need 1,398 samples for every average approximation.

4.3 Propagation Models

What are propagation models? They are mathematical prediction formulations derived from empirical data collected in a given condition and environment. Hopefully, when these models are applied to similar conditions and environment, they will predict the propagation with reasonable accuracy.

Why do we need to use propagation models?

1. Data collection through drive testing and recollection every time it is needed is expensive.
2. Drive testing is time consuming.
3. Drive testing is inconvenient.

Choosing a prediction tool:

1. A prediction tool is a quick and reasonably accurate estimation of the performance and build-out verification, provided it is tweaked well.
2. Not all prediction tools are created alike. Each tool in the market has pros and cons.
3. Some have a frequency limitation outside of which they are grossly inaccurate.
4. Others have distance limitations outside of which they introduce errors. This points to the fact that propagation prediction is very nonlinear and depends on a lot of factors, not just frequency and distances.
5. While predicting path losses is the easiest thing to do, modeling obstruction and clutter and predicting diffractions around those obstructions can be challenging.
6. A common denominator of all prediction tools is the resolution of terrain data. No type of prediction tool can be accurate without the help of a reasonably high resolution of terrain data. The rule of thumb is a 30-m resolution. Any higher resolution data, while it can make a little bit of difference in the accuracy of the prediction, will not probably justify the higher cost.
7. Some prediction tools have simple algorithm engines, while others are complex. Most are based on empirical data underlying the mathematical formulations, while others use wave or ray tracing methodology, thereby relying heavily on accurate clutter modeling and very high-resolution terrain data—as high as 10m.
8. While coverage prediction is easier, capacity prediction is harder, especially for a dynamic load-dependent technology, such as CDMA. In this case, the capacity-predicting algorithm needs an additional component to compute the random distribution of subscribers, computation of network load and power, redistribution of subscribers, and recalculation of loads and powers, until a steady state is reached at which time the load at a predetermined level can be computed. For such a computation, typically a Monte Carlo type simulation is used.

9. In technologies such as WiMAX, dynamic computation of coverage and capacity is hard, especially in light of the fact that MIMO and Smart antennas can complicate prediction. On the capacity side, modeling now is based on packet rates and delay, as opposed to a static Erlangs computation. Erlang C is the more appropriate model.

The question of accurate prediction is an attempt to answer one or two main questions: How complete is my coverage (signal levels at various points) in a given area and at what confidence level? How many customers can I support?

In Section 4.3.1, we shall discuss the fundamental types of propagation models.

4.3.1 Free Space Path Loss Model

The free space path loss model is considered a very fundamental prediction tool, compared to all others, and as such it serves as a baseline. A prediction model considered at its most basic, the free space path loss model takes into account only distance and frequency.

$$PL_f = 32.4 + 20\log_{10} d_{\text{km}} + 20\log_{10} f_{\text{MHz}}$$

where PL_f is free space path loss in dB, d_{km} is distance in kilometers, and f_{MHz} is frequency in MHz.

Taking only the distance component, the others remaining constant, the free space path loss model says that for every factor of 10 (km) that the receiver is away from the transmitter, the loss varies by 20 dB.

Similarly for the frequency component, for every factor of 10, the path loss gain is higher or lower by 20 dB.

Let's take an example in the WiMAX frequency range of 3.5 GHz, or 3,500 MHz, at a distance of 1.5 km.

PL(free space) = 32.4 + 20log(1.5) + 20log(3,500) = 32.4 + 3.5 + 70.9 = 106.8 dB.

For frequency corresponding to 2,500 MHz, the free space path loss is 3-dB lower (103.8 dB). This result indicates that I have double the output of my

Table 4.1

Distance-Dependent Path Losses in dB

	$PL_{f(d)} = 20\log_{10} d_{\text{km}}$			
d_{km}	1	10	100	1,000
$PL_{f(d)}$	0	20	40	60

Table 4.2
Frequency-Dependent Path Losses in dB

	$PL_{f(f)} = 20\log_{10} f_{MHz}$			
f_{MHz}	10	100	1,000	10,000
$PL_{f(f)}$	20	40	60	80

transmitter power in 3,500 MHz than in 2,500 MHz in order to overcome frequency-related losses.

4.3.2 The Okumura-Hata Model

The Okumura-Hata model (also called the Hata model) is an empirical model developed by Hata and derived from a technical report written by Okumura. The Okumura-Hata model is widely used in predicting radio propagation in cluttered areas. It calculates attenuation, taking into account the percentage of buildings in a path, as well as natural terrain features.

The Okumura-Hata model for an urban area is given by [4]:

$$PL = 39.55 - 26.16\log f - 13.82\log h_{te} - a(h_{re}) - \left(44.9 - 6.55\log(h_{te})(\log d)\right) \tag{4.5}$$

where:

PL: Path loss in urban areas, given in dB;

f: frequency (MHz), from 150 MHz to 1,500 MHz;

h_{te}: BS effective antenna height above ground (m) in the range of 30–200m;

d: distance (km), between transmitter and receiver;

$a(h_{re})$: mobile antenna height correction factor;

$a(h_{re})$: $(1.1\log f - 0.7)\, h_{re} - (1.56\log f - 0.8)$, medium-sized city;

h_{re} = mobile station antenna height above ground (m) in the range of 1–10m;

For large cities:
$a(h_{re})$: $8.29(\log(1.53 h_{re}))^2 - 1.1$, if f is 150–200;

$a(h_{re})$: $3.2(\log(11.75\, h_{re}))^2 - 4.97$, if f is 200–1,500;

Correction for a suburban area:

$$PL_{SU} = PL - 2\left(\log_{10}\frac{f}{28}\right)^2 - 54 \tag{4.6}$$

where PL_{SU} is the path loss in a suburban area, while PL is the path loss for an urban area as shown above.

Correction for open area:

$$PL_{\text{open}} = PL - 4.78(\log f)^2 + 18.33\log f - 40.97 \tag{4.7}$$

where PL_{open} is the path loss for an open area.

One can notice that BS heights below 30 and radii of less than 1 km will not be accurately predicted. In environments in which antenna heights have to be kept below 30m and cell sites closer together, within 1 km, for capacity needs (as in pico cells), this is not the tool to use, and users are forewarned.

A word or two about empirical models: Empirical models measure signal response at a frequency per distance for a specific environment and are rather accurate for that location. However, it is expensive to duplicate the models again and again for other locations. The labor and equipment cost is expensive. Processing the data is time consuming. It is also hard to generalize frequency and environment.

4.3.3 PCS Extension to the Hata Model

PCS extension Hata, also known as Cost 231, is another version of the Hata model, with the only difference being that the original Hata model has new frequency intervals, namely, 1,500–2,000 MHz. The PCS extension is given by:

$$\begin{gathered} PL = 46.3 - 33.9\log f - 13.82\log h_{te} - a(h_{re}) - \\ (44.9 - 6.55\log(h_{te})(\log d)) + C_{Env} \end{gathered} \tag{4.8}$$

where

PL: Path loss given in dB;

f: frequency (MHz), from 1,500 to 2,000 MHz;

h_{te}: BS effective antenna height above ground (m) in the range of 30–200m;

d: distance between transmitter and receiver, 1–20 km;

h_{re}: Mobile station height of 1–10m, a function of the size of the coverage area;

$a(h_{re})$: as defined above in the Hata model;

C_{Env}: Environmental correction factor

= 0 dB for a medium-sized city and suburban areas;

= 3 dB for metropolitan centers.

4.3.4 Longely-Rice Model

Also known as the irregular terrain model (ITM), the Longely-Rice (LR) model is a propagation-prediction tool for path losses of mainly point-to-point (PTP) links.

LR works in the frequency ranges of 40–100 GHz (some 20 MHz–20 GHz) and distances less than 2,000 km.

LR has two parts: a model for predictions over an area and a model for PTP link predictions. What LR does not correct for is the effect of buildings and foliage, nor does it take multipaths into account. The model takes the following inputs into account in order to compute a path loss:

- Conductivity of the ground over which the signal propagates;
- Dielectric constant or the relative ground permittivity;
- Refractivity of the atmosphere (effective Earth curvature, K);
- Climate zone code, such as equatorial, tropical, or desert, and so on;
- Antenna polarization, vertical orhorizontal;
- Receiver height;
- Situation variability (between 1 and 100%, with 50% being the average);
- Time variability (between 1 and 100%);
- ITM mode (broadcast or individual);
- Urban clutter factor.

Table 4.3 shows the input parameters just listed, with their corresponding typical values.

4.3.5 Indoor Propagation Model

Indoor environment propagation is dominated by both LOS and NLOS (Rician distribution) signals. The LOS is due to the short distances from the transmitter to the receiver, while the NLOS is due to the reflections of signals off nearby walls, partitions, furniture, other objects, or even people.

At times the transmitter has to traverse multiple floors, walls, or partitions. Indoor reflections can be serious anomalies that can literally wipe out any usable

Table 4.3

Longely-Rice Typical Input for Path Loss Calculation

Refractivity (*N*-units)	300	Receiver height AG:	9m	
Dielectric constant	15			
Conductivity (Siemens/m)	0.005	Antenna polarization		
Permittivity	15.0	Horizontal ☐	Vertical ☒	
Time variability	10%		Climate→→	☒ Desert
Situation variability	50%			☐ Tropical
				☐ Equatorial
TM Mode →→	☒ Broadcast			and so on
	☐ Individual			

signals, if the proper technology is not selected to at the beginning. In our discussion earlier in connection with selective fading and delay spread, it was pointed out that you need a low data rate and a bigger symbol period to resolve the shorter delay spreads inherent in the in-building (or dense urban) environments.

Here is a simple indoor propagation model:

$$L_p(\mathrm{dB}) = \beta(\mathrm{dB}) + 10\log_{10}\left(\frac{r}{r_0}\right)^n + \sum_{p=1}^{P} WAF(p) + \sum_{q=1}^{Q} FAF(q) \qquad (4.9)$$

r : distance between transmitter and receiver;

β(dB): loss at reference distance r_o;

r_o: nominal reference distance (typically 1m);

WAF(p): wall-attenuation factor, for p walls;

FAF(q): floor-attenuation factor, for q floors;

n: ≈ 2 for close distances, larger for greater distances.

Table 4.4

Wall Material Losses

Material*	**Loss at 900 MHz**	**Loss at 1,700 MHz**
Plaster wall	≈5 dB	≈9 dB
Concrete wall	≈10 dB	≈17 dB

*You have to see a more exhaustive list of material-related losses.

More accurate when p and q are small, the model neglects *angle of incidence* and effect of distance on n.

The Stanford University Interim (SUI) model, chosen for WiMAX propagation studies in this book, is covered in Section 6.3.

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5

Communication Channel Characterization

5.1 Gaussian Noise and Line-of-Sight and Nonline-of-Sight Paths

The communication channel is largely affected, for the most part, by multipath and noise. In the additive white Gaussian noise (AWGN) channel, the signal is degraded by thermal noise and is typical of space and wire transmissions. On the other hand, Rayleigh fading characterizes the absence of a LOS path and more closely models the outdoor mobile environment. Rician distribution characterizes both LOS and NLOS paths and model an indoor, smaller cell, or open environment.

The channel can be characterized by a parameter M:

$$M = \frac{P_{LOS}}{P_{NLOS}} \tag{5.1}$$

where M is a ratio of P_{LOS}, power in a LOS path, to P_{NLOS}, power in an NLOS path.

When $P_{LOS} = 0 \Rightarrow M = 0$, implying Rayleigh fading (distribution). And if $P_{NLOS} = 0 \Rightarrow M = \infty$, implying the channel is AWGN. Anywhere between those two values is a combination of the two, with some LOS and some NLOS, a typical wireless communication channel characteristic. Refer to Figure 5.1 [1].

5.2 Characterizing the Radio Environment

A typical wireless mobile receiver can experience one of the following radio environments:

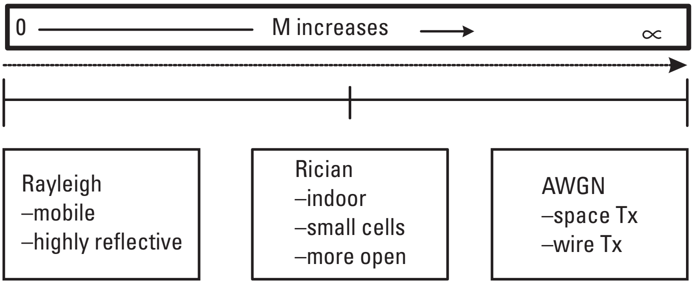


Figure 5.1 Channel classification summary.

1. Distance, *d*, dependent loss, of $\frac{1}{d^2}$ for free space and $\frac{1}{d^4}$ for a typical cluttered environment. *d* is a far field on the order of many wavelengths of the signal.
2. At distances on the order of a fraction of a wavelength, due to a multipath fading, received signal, can vary many dBs. Multipath fading, a random variable, can be modeled with a Rayleigh distribution, especially in outdoor environments. More discussion on Rayleigh distribution is provided later.
3. At distances over many wavelengths, received power to a receiver in motion can experience shadow fading. Shadow fading is a random variable modeled by log normal distribution.

Let's generalize received power as P_r. P_r comprises the following:

- Transmit power;
- Shadow fading;
- Multipath fading;
- Distance dependent loss (over and above the reference distance);
- Transmit antenna gain;
- Receive antenna gain;

and is given by:

$$P_r(\text{dB}) = -10\log_{10}\varphi - 10\log_{10}\vartheta - 10\log_{10}\left(d^{\alpha}\right) + 10\log_{10}P_t + G_t(\text{dB}) + G_r(\text{dB}) + 10\log_{10}P_{dref} + L_{other} \tag{5.2}$$

where φ = a shadowing effect modeled as $\varphi = 10^{\frac{x}{10}}$, where x is a Gaussian normal distribution with 0 mean and variance σ^2, and random variable,

ϑ = multipath effect;

α = path loss exponent;

P_{dref} = received power at a referenced distance;

G_t = transmit antenna gain;

G_r = receive antenna gain;

L_{other} = other losses.

Regarding log normal fading, we stated that φ, as defined in (5.2), is a shadowing effect modeled as a lognormal random variable:

$$\varphi = 10^{\frac{x}{10}} \tag{5.3}$$

The decimal form x, whose logarithm is given by:

$$x = 10\log_{10}\varphi$$

and is normally distributed and called log-normally distributed. This is the form used in the link budget and calculation involving decibels. (Refer to the following extract.) So, in effect, we are back to the random variable x modeling the shadowing effect.

A normal distribution (also called a Gaussian distribution) is defined by two parameters: mean and variance or standard deviation. Note that a standard normal distribution has 0 mean and 1 standard deviation.

> Explaining Log Normal Fading Through the Central Limit Theorem:
>
> When taking samples, regardless of the population the samples are taken from, the mean of random samples approaches a normal distribution for a large sample size. For a more concrete example, if you flip a fair die many times, each outcome has a probability of 1/6, and the distribution is flat all the way for outcomes 1, 2, 3…6. The mean is (1 + 6)/2 = 3.5. Now, if we instead flip two dice, one after the other, and sum the result (alternatively we can flip the two dice at the same time, adding the outcome) the result would be 2, 3, 4…12. The probability of each outcome now is different. [For example, an outcome of a 5 could be (3,2), (2,3), (4,1), (1,4), that is, 4 outcomes out of a possible 36 outcomes (6·6, for two dice).] Hence, the probability of distribution from 2, 3, 4, 5…12 is *not* flat, but in fact starts small, peaks, and becomes smaller and smaller again. It follows the pattern of a bell or a normal curve. The mean, in the middle, is 7. Now, if we flip 3

dice, one after the other, and sum the result, the possible outcomes will be even more like a bell curve. This is one concrete example of a central limit theorem.

The signals arriving at the receiver are random, multiple, and independent (from a Gaussian distribution, each signal is independent to the other). The sum of these signals, under some constraints, converge to a Gaussian form. Since the mobile subscriber station (SS) is changing position constantly, the shadowing or losses due to obstructions change and, hence, each attenuation changes. Let's assume that there are *n* number of such attenuations, or random variables, converted to dB, that accumulate to make up a normal Gaussian distribution. Thus:

$$x = \sum_{i=1}^{n} 10 \log_{10} \theta_i = 10 \log_{10} \varphi \tag{5.4}$$

where θ_i is the *i*th attenuation variable. By central limit theorem, (5.4) is lognormal.

5.3 Multipaths and Doppler Shifts

For this discussion, refer to Figures 5.2–5.5.

Signals with no clear and dominant LOS path arriving at a mobile receiver, such as independent, random variables, via multiple paths, constitute Rayleigh fading.

Again, such signals, by the central limit theorem, converge to a Gaussian form. For a Gaussian signal, we can thus recognize that the envelope random variable of the signal in this type of multipath represents a Rayleigh distribution.

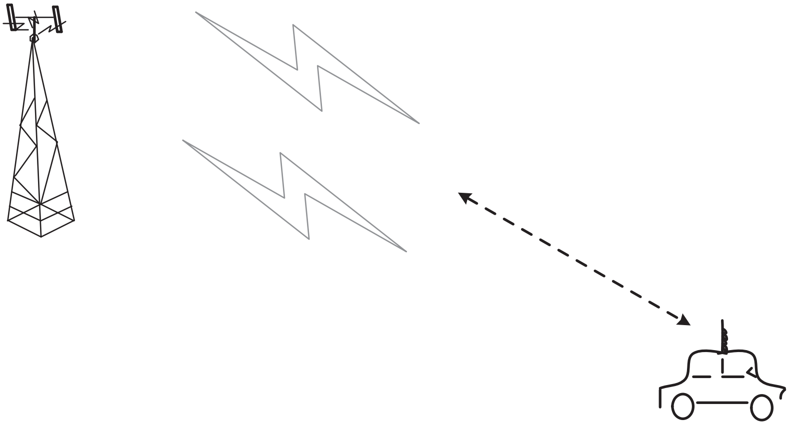


Figure 5.2 Static path loss.

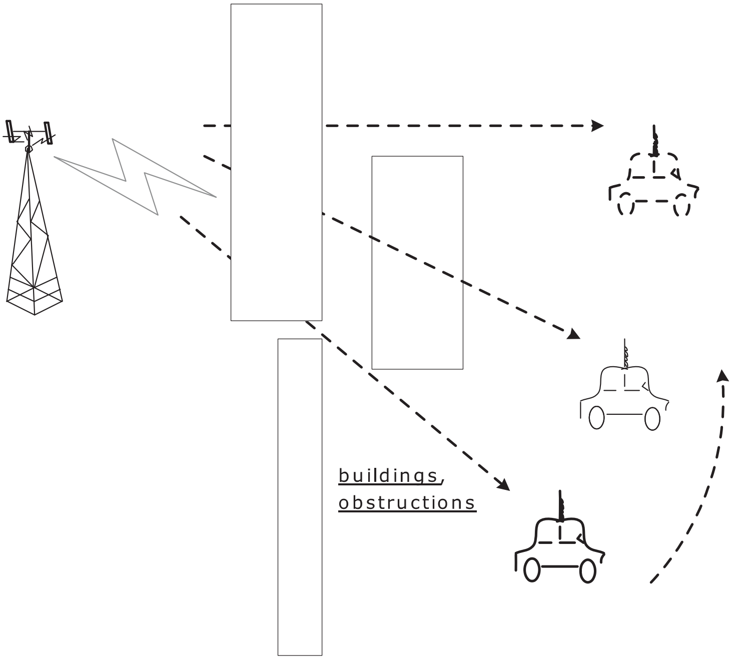
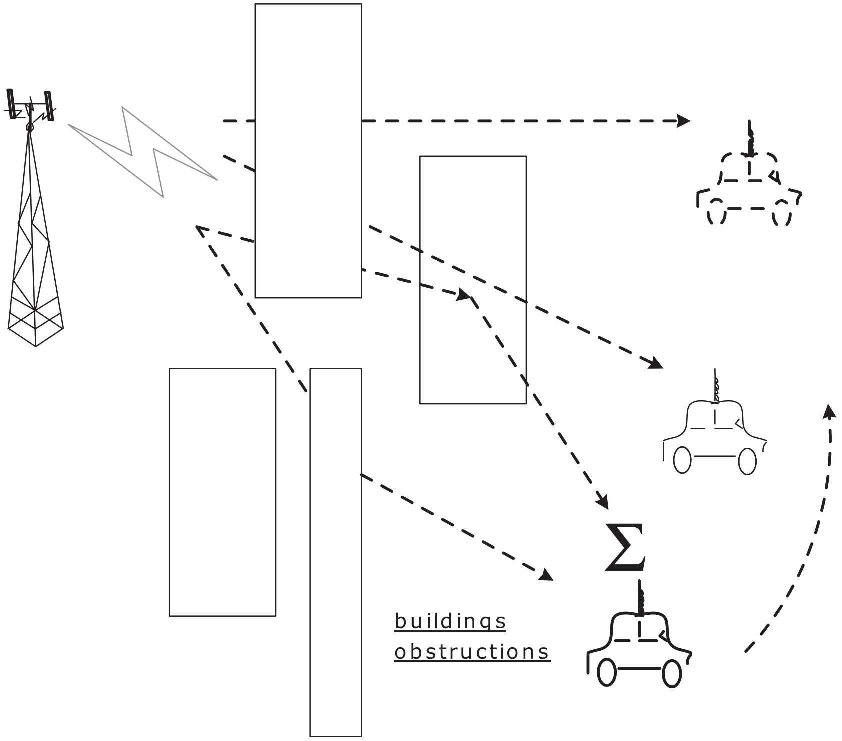


Figure 5.3 Lognormal fading.

Rician fading is similar to Rayleigh fading, with only one exception—it has a LOS path or straight path dominant component between the transmitter and receiver.

As pointed out, the *n* random signals arriving at the receiver via different paths, or multipaths, are linear and time-invariant random variables, *x*. However, when a vehicle starts to move, the signals become time-variant (time-dependent) random process variables, $x(t)$. As the receiver moves from time t_1 to t_2, the environment effectively changes. The *n* random variable at $x(t_2)$ is different from the random variable $x(t_1)$. It is noteworthy to mention the similarity of the signals at t_1 and at t_2, assuming t_1 and t_2 are *sufficiently* close [2]. One means to predict the similarity is to make use of autocorrelation analysis. See the following extract.

> Autocorrelation and Wide-Sense Stationary:
>
> Autocorrelation is a measure of the similarity of a random process, just as variance is a measure of randomness. Autocorrelation is given by $R_x(t_1, t_2) = E\{x(t_1)x(t_2)\}$, where the right term is the expectation or



Multipath fading - NLOS (distribution: Rayleigh)

Figure 5.4 Multipath fading.

average of the products in the bracket

If $x(t_1)$ and $x(t_2)$ are uncorrelated, then

$$R(t_1, t_2) = E\{x(t1)\}E\{x(t2)\}$$

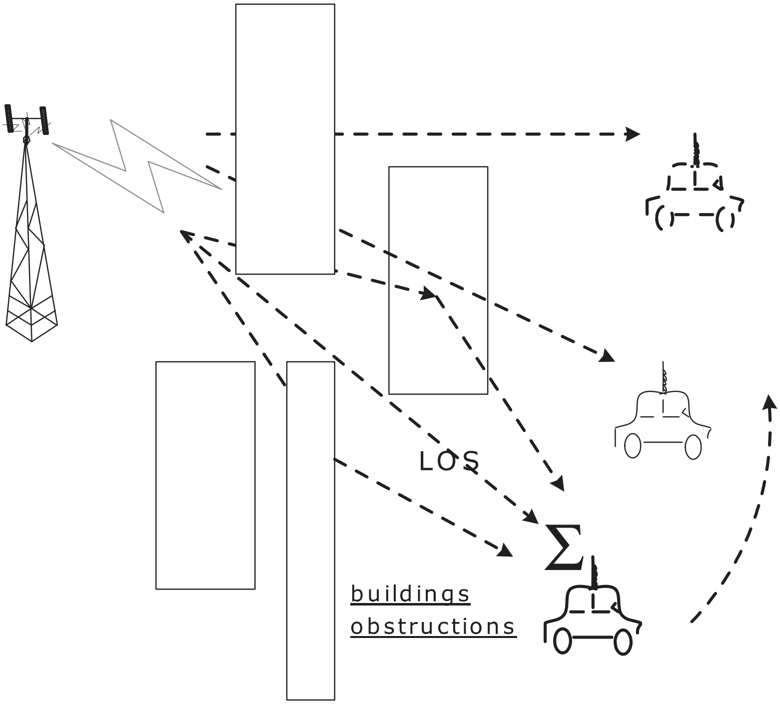
There is a means of simplifying the autocorrelation function—wide sense stationary (WSS) autocorrelation. WSS is origin-of-time invariant and depends on the difference in time only. The autocorrelation for WSS is:

$$R_x(\tau) = E\{x(t+\tau)x(t)\} = E\{x(t)x(t+\tau)\} = R_x(-\tau)$$

where $\tau = t_1 - t_2$, the difference in the two times as τ increases, the autocorrelation becomes the product of the mean values. Thus, $E\{x(t).x(t+\tau)\}$ are uncorrelated, implying the frequency domain of $x(t)$ is influenced by high-frequency components.

A random process, $x(t)$, is WSS if:

1. The mean is constant, $m_x = E\{x(t)\}$.



Multipath fading - both LOS and NLOS
(distribution: Ricean)

Figure 5.5 Direct and multipath propagation.

2. The autocorrelation is a function of only the time difference, $(t_1 - t_2)$, $R_X(t_1, t_2) = R_X(t_1 - t_2)$.

Clearly, in our case, each random variable $x(t_1)$ and $x(t_2)$ is a Gaussian distribution with a mean of 0.

Secondly, the autocorrelation can be viewed to depend on time differences, for small τ.

So, as the vehicle changes position and moves in time from t_1 to t_2, provided the difference in time is small, the signals are correlated. As the time difference τ increases, they become more and more uncorrelated.

Note that the two uncorrelated random variables are not necessarily independent, but Gaussian random variables are independent. By the same token, any independent random variables are uncorrelated.

WSS autocorrelation is significant, as the Fourier transform of the autocorrelation is the power spectral density (a measure of power in a frequency domain) of the random process.

It is important to point out that the speed of the vehicle, equivalent to changing positions fast, contributes to a pronounced Doppler shift that leads to the spilling of the frequency components outside the coherence bandwidth. More discussions regarding Doppler shifts and a concrete example are given in Section 2.5.

5.4 Nyquist and Shannon-Hartley

The communications engineer aims to optimize his design for an efficient communication system, taking into account various competing goals. Ideally, the designer proceeds with an eye toward a system with the highest data rate possible and a minimal bit error rate, using minimal power and spectrum. In so doing, the designer endeavors to offer reliable service to as many users as possible, with as minimal interference and as high reliability and quality as possible. The engineer also aims for minimal complexity and cost. These goals are a tall order in the face of a technology replete with fierce competition.

5.4.1 Noise, Noise Floor, and Interference Definitions

Let's define some key terminology that will enable us to characterize a competitive and efficient system.

Thermal noise density, N_{th}, is given by $N_{th} = \mathrm{k}T$, where k is Boltzmann's constant ($1.38 \cdot 10^{-23}$ J/K) and T is the temperature in Kelvin, K.

Total noise for a given bandwidth, B, is $N = \mathrm{k}TB$, where B is given by Hz.

Receiver noise floor is given by $NoiseFloor = \mathrm{k}TBNF$, where NF is noise figure of the receiver.

The signal-to-noise ratio is $\frac{S}{N} = \frac{E_b}{N_o} \cdot \frac{C}{BW}$, where the ratio E_b/N_o is bit energy to noise spectral power density, BW is bandwidth, and C is carrier power.

Carrier-to-noise ratio is given by C/N = carrier level at receiver input/noise floor of the receiver.

Interference plus noise in decibels is given by $I + N$.

Note: Convert each component (I and N) to its decimal counterpart first, add them, and convert them back to decibels.

5.4.2 Nyquist Minimum Bandwidth

In theory, only *Rs*/2 is required to transmit 1 symbol/s. (where *Rs* is the symbol rate.) However, practically, *Rs* Hz is required to transmit 1 symbol/s, that is, 1 symbol/s/Hz.

Given M symbols [1] corresponding to k bits, that is, $M = 2^k$ symbols for a fixed bandwidth (BW), as k increases, data rate, R, increases. There comes a point, however, where either the BW or SNR has to be increased. Otherwise, the symbols cannot be differentiated due to AWGN. Note that we are not taking ISI into account.

Nyquist can be summarized as follows:

$$C = 2BW \log_2 M \tag{5.5}$$

where M is the number of discrete signals, C is the data rate, and BW is the bandwidth. For example, for $M = 2$, the data rate is twice the BW. For $M = 8$, implying 3 bits, the data rate is six times the BW. We cannot keep increasing M, as noise and other impairments limit the distinction of each signal.

5.4.3 Shannon-Hartley

The Shannon-Hartley capacity theorem states that

$$C = BW \log_2 \left(1 + \frac{S}{N}\right) \tag{5.6}$$

where S/N is the signal-to-noise ratio, BW is bandwidth, and C is the capacity in bits/s. Implying the data rate, $R \leq C$, approaching C by finding the right code and setting a limit on R.

Determining the Shannon Theoretical Limit of $\frac{E_b}{N_o}$

Taking the Shannon capacity formula,

$$C = BW \log_2 \left(1 + \frac{S}{N}\right)$$

Dividing both sides by $\frac{C}{BW}$ gives bits/s/Hz, a measure spectrum efficiency.

Now, continuing with defining $\frac{S}{N}$, where $\frac{S}{N} = \frac{E_b}{N_o} \cdot \frac{C}{BW}$; then $\frac{C}{W} =$

$\log_2\left(1+\frac{E_b}{N_o}\cdot\frac{C}{BW}\right)$.

Using the identity as suggested in [3], $\lim_{x\to 0}(1+x)^{\frac{1}{x}}=e$ with $x=\frac{E_b}{N_o}\left(\frac{C}{BW}\right)$ and solving for $\frac{E_b}{N_o}$ gives $\frac{E_b}{N_o}=0.693=-1.59$ dB, which is the Shannon limit when $\frac{C}{BW}$ approaches 0. As $\frac{C}{BW}\to 0$, implying that by improving the coding scheme to a point $k\to\infty$ (*k*-bit, symbols, and BW) it is possible, theoretically, to achieve E_b/N_0 of –1.6 dB. This is only a theoretical limit. Refer to Figure 5.6 for a graphical illustration. Please keep in mind the power and bandwidth efficiency tradeoffs discussed in Section 3.2.

Please refer to Figure 5.6. It is impossible to achieve this theoretical limit: as *k* increases without bound, so will the *BW*. However, an improved implementation technique will approach that theoretical limit. Coding and modulation technique will help in reducing transmit power and conserving bandwidth. For

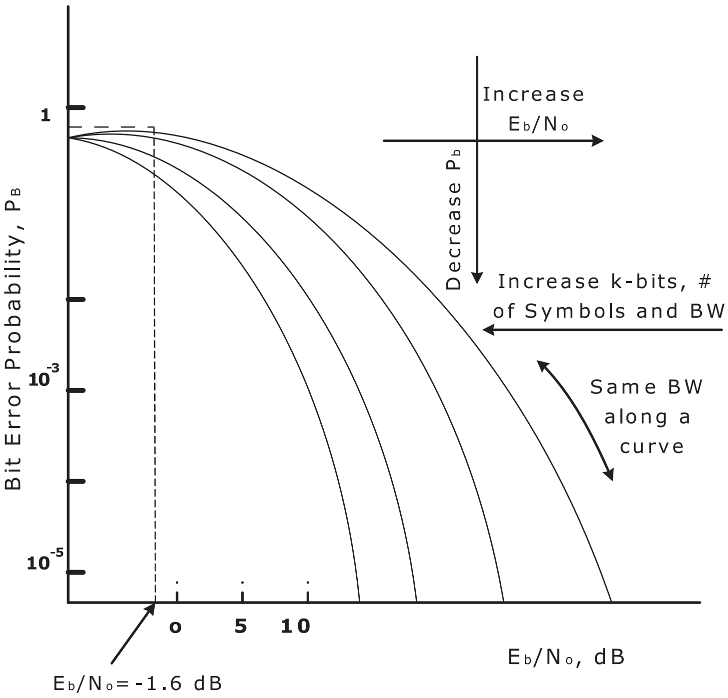


Figure 5.6 Bit-error probability related to E_b/N_o.

example, 64 QAM is an efficient modulating technique that uses $M = 2^6$ where $k = 6$.

5.5 Separation of Colocated Antennas

5.5.1 Vertical and Horizontal Separation

Frequency separation or distance separation is effective in mitigating interference for colocated antennas. At times and when necessary, both geographic and operating- frequency separations must be used to avoid or solve interference problems.

Many times different carriers find themselves operating in spectrums close to each other, and because of the scarcity of spectrum, operators occupy contiguous frequencies. Due to the onerous requirements to locate communication antennas in certain neighborhoods, operators invariably are forced to colocate on the same tower as their competitors. The frequencies of the rival operators could be very close as well. That means that there is a real squeeze in both spectrum availability and geographic location. Because of these reasons, designers have to adhere to a guideline for how far they should separate the colocated antennas to minimize interferences. Refer to Figure 5.7 for a graphical illustration of horizontal and vertical separation.

To do away with interference between two operators, you need sufficient isolation loss, separation in spectrum or geographic distance, or a combination of both. For example, if you take the free space path loss:

$$PL = k + 20\log_{10}(d_{\mathrm{km}}) + 20\log_{10}(f_{\mathrm{MHz}}) \tag{5.7}$$

All other things being equal, here is how to get a head start with calculating the loss necessary for isolation. A 10-km separation will result in a 20-dB loss [not adding the k and frequency loss, as shown in (5.7)]. Similarly, a 100-km separation will result in a 40-dB loss, and so on. Generally, that is a 20-dB-per-decade loss. Also similarly, a 10-MHz separation in frequency will result in 20-dB loss [not adding the k and distance loss, as shown in (5.7)] and a 100-MHz separation in frequency will result in a 40-dB loss.

Both frequency and distance separation are governed by the *same* law, namely, 20 dB per decade. Every time you increase the distance or frequency by a factor of 10, the loss increases by 20 dB. This can be deduced by looking at the coefficients in the path loss formula.

Here is one way to optimize the two separations (frequency and distance). Let's say you can separate the Operator A site from the Operator B site by only 1 km in geographic distance. Let's also assume you can separate the two operators' frequencies by 10 MHz. Thus, the isolation loss gained (all other things being equal), is 20log(10 km) + 20log(10 MHz)= 40-dB isolation loss (+ *k*, of course).

Now, if the two operators share the same tower, the geographic isolation loss component is diminished or avoided altogether. Additionally, if there are antenna gains toward each other, then the isolation loss is further diminished. It also matters whether the antennas are separated horizontally or vertically. Vertical separations offer better losses, owing to the fact that the antenna patterns are on different planes. Usually, whether to use vertical or horizontal separation is predicated by the availability of space on the tower or building. Further, if the antennas are slanted, having both vertical and horizontal components, then contribution of each is resolved by taking the angle into account. The next step is to use the following formulas and then add the results of the two.

Figure 5.7 shows a typical separation of distance on a tower top.

The following are formulas that combine distance and frequency losses for both vertical and horizon separation situations [4].

Vertical separation loss (D_v) is given by:

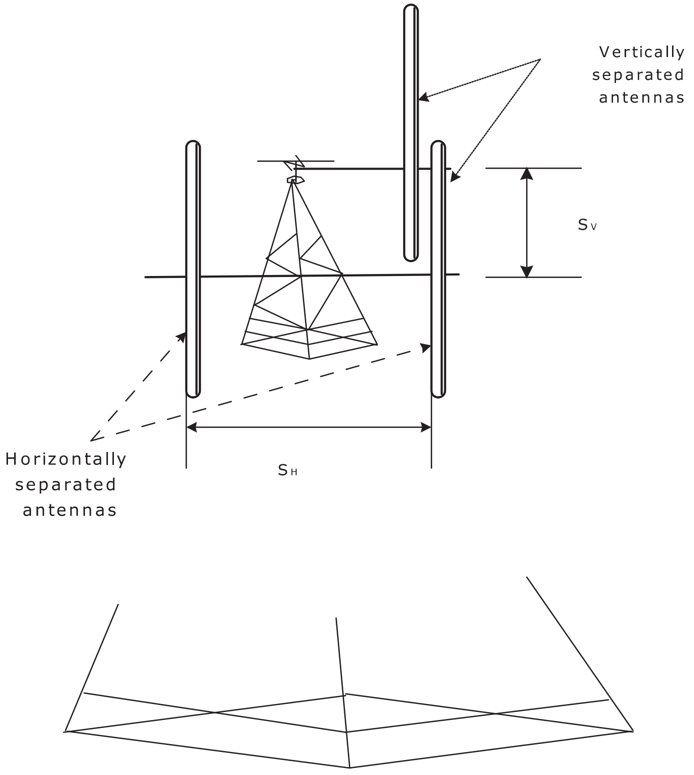


Figure 5.7 Antenna separation.

$$L_v = 28 + 40\log_{10}\left(\frac{D_v}{\lambda}\right) \tag{5.8}$$

Horizontal separation loss (D_h) is given by:

$$L_h = 22 + 20\log_{10}\left(\frac{D_h}{\lambda}\right) - (G_1 + G_2) \tag{5.9}$$

where, in each case,

$$\lambda = \frac{c}{f}$$

G_1, G_2, are antenna gains toward each other

$$c = 3 \times 10^8 \frac{m}{s} \text{ (speed of light)}$$

Bear in mind that separations at an angle can be resolved to their respective vertical and horizontal components before applying the loss relationships above. Then, of course, the two groups are added.

5.6 Intermodulation

Intermodulation (IM) is a severe interference.

There are three main sources of IM interferences:

1. Within receiver;
2. From transmitter;
3. Related to antennas, systems, or cables.

A receive IM results when two or more signals of different frequencies passing through a nonlinear device, such as a mixer, create strong third-, fifth-, and seventh-order IM products of another frequency that can fall within a desired receive frequency. The third order is the strongest and most destructive interference. Figures 5.8 and 5.9 illustrate these facts.

As shown in Figures 5.8 and 5.9, a nonlinear device is a major source of IM. The multiplier is usually a suspect, since it creates products of other

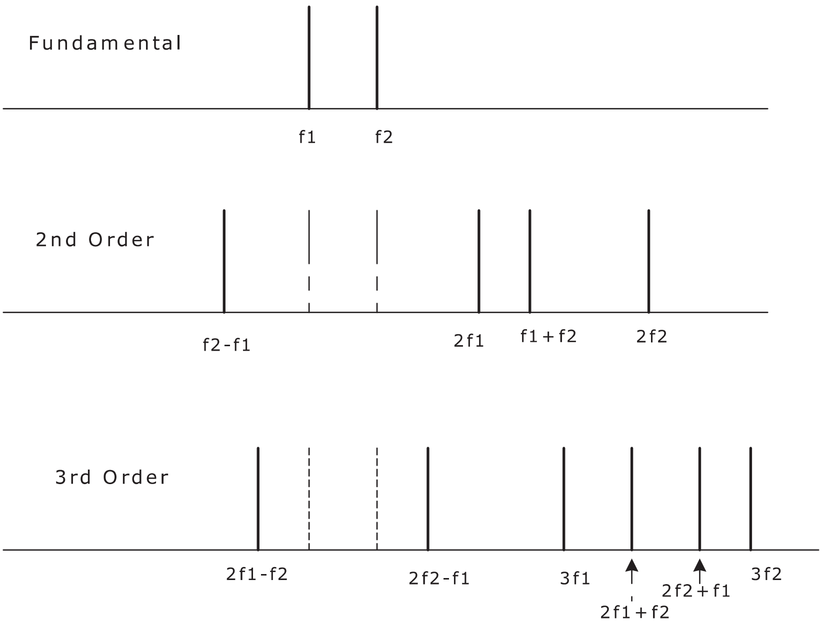


Figure 5.8 IM products.

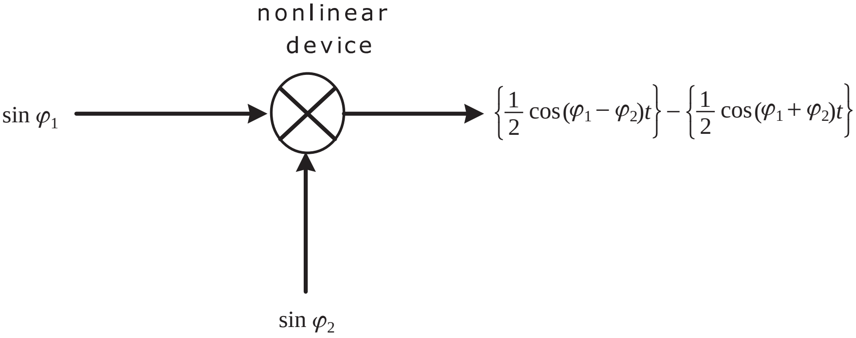


Figure 5.9 Two signals passing through a multiplier.

frequencies as the sum and differences of the original sinusoids similar to the IM products shown in Figure 5.9.

5.6.1 Strategies to Avoid, Troubleshoot, and Mitigate IM

- Identifying all the possible frequencies: In an area where there are numerous interference sources, computer software should be used to determine all the possible second- and third-order products. If, for example, the calculated frequency is the same as the receiving frequency, one has to yet determine whether the IM product is generated at the input of the receiver or if it came from the transmitter's mixers.

- Isolating: Generally, an attenuator installed in the antenna cable of the receiver can be used to isolate the source of IMD, whether it is from the transmitter or receiver.
- Before site development: Before developing a site location and investing $100,000 or more, it is a good idea to sweep the frequency environment around the clock to identify IM products. One should sweep the radio spectrum between 25 MHz and 7 GHz for a period of 24 hours.
- IM mitigation techniques:
  - Transmitter shielding to reduce coupling between transmitters;
  - Correct antenna installation;
  - Enough antenna isolation (vertical and horizontal separation). Vertical separation has greater coupling losses than horizontal separation. Refer Section 5.5 for quantitative analysis and more information on this topic.
  - Proper grounding, especially of the transmitter. The larger the diameter of the grounding copper, the better the protection is going to be.
  - Proper antenna-to-transmitter matching.
  - Filtering at the transmitter and receiver, which attenuates the amplitude of the interfering signals.
  - Use of isolators, circulators, and a harmonics filter.
  - Passive IM originates in nonactive devices (antennas, for example), loose joints, and dissimilar materials. Use DIN connectors.
  - To isolate whether IM originated from an outside receiver or from within the receiver, install an attenuator on the receive side, right after the antenna. Then, if the IM level, as seen on the spectrum analyzer, is the same as before the attenuation was installed, more than likely the dominant component originated from within the receiver. In that case, it might originate from the passive elements or from local oscillators.
  - Calculation of IM. Unless it is below the receiver threshold, the following must be examined for better isolations: antenna separations, filters, and isolators.
- Typical power-level calculation of transmitter-originated IM, which can occur from one of the following two:
  1. Frequencies of external origin mix in the receiver to create IM.

2. Frequencies originating both externally and those originating from the harmonics of the receiver's local oscillators combine to create IM.

- Two steps in calculating transmitter IM:

1. Determine the most likely transmitter frequencies that contribute to the receiver's band. For third-order IM from TX_1, TX_2, and RX, determine if $(2TX_1 - TX_2)$ or $(2TX_2 - TX_1)$ are near in the RX band.

2. Calculate the power level at RX coming from either $(2TX_1 - TX_2)$ or $(2TX_2 - TX_1)$, depending on which one is in the RX band, as calculated in 1. Then,

$$P_{TX} - L_{TX1-TX2} - L_{filter} - L_{TX-RX}$$

where

P_{TX} is the transmit power level (of, say, TX_1, taking the worst-case scenario)

$L_{TX1-TX2}$ = separation loss between two transmitters

L_{filter} = two way filter in TX_2

Compare the power level calculated to the receiver noise floor (total noise + NF of the receiver). The calculated power should be well below the noise floor. Otherwise, it is indicative of a presence of significant IM products.

5.7 WiMAX Versus Satellite Interference Consideration [5]

The C-band is in the range of 4–8 GHz. A typical C-band satellite uses:

- 3.7–4.2-GHz downlink;
- 5.925–6.425-GHz uplink.

The initial profile for WiMAX uses the upper U-NII/ISM band of 5.725–5.850 GHz (refer to spectrum for WiMAX in Figure 1.2).

Recent testing showed that fixed WiMAX transmissions could cause interference to a satellite digital signal as far away as 12 km to fixed satellite services (FSS) operating in the C-band. Hence, the study concluded that FSS antennas cannot coexist with WiMAX systems in ranges from 50 km to 200 km, depending on the terrain and WiMAX power output levels [5].

5.8 Interferences

In addition to the channel challenges discussed, interferences, such as adjacent, cochannel, and far-near effects, are critical.

5.8.1 Adjacent Channel Interference

Even though adjacent channel interference is known more in frequency division multiple access (FDMA) and time division multiple access (TDMA), to get a sense of what is involved, it is quantified here. Owing to the contiguous nature of frequencies, when they have to be used together, an efficient filter is the main defense against mutual interference, in the absence of a guard band and with limited geographic separation [6].

Given by

$$Loss_{adjacent} = k \log_2\left(\frac{f_2}{f_1}\right) = \frac{k}{0.3}\log_{10}\left(\frac{f_2}{f_1}\right) \tag{5.10}$$

where k is the dB/octet slope of filter.

Equation (5.10) says that the larger the filter slope, k, and the greater the separation of f_2 from f_1, the greater the protection from adjacent channel interference.

5.8.2 Distance Path Loss of Geographic Separation

The distance path loss of geographic separation is

$$Loss_{cochannel} = 40\log_{10}\left(\frac{d_1}{d_2}\right) \tag{5.11}$$

where $d_1 > d_2$ and both are distances from two cochannel sources to the base station [6].

5.8.3 Near-End-Far-End

A near-far problem occurs any time two signals are received from two locations. This can be BS ←mobile → BS or SS ← SS → SS. Normally, power control mitigates or eliminates the near-far problem by effectively balancing the powers.

While distance separation aids in signal interference, there is an exception in which distance separation can be detrimental to communication. It is called the *near-end-to-far-end ratio.*

$$\text{Near-end-to-far-end ratio} = \text{path loss due to } d_2 \text{ (near-end)} / \text{path loss due to } d_1 \text{(far-end)} \quad (5.12)$$

This is because the stronger (or the closer in distance, d_2) will *mask* or dominate the weaker (the further in distance or d_1) signal.

5.8.4 Antenna Directions

If the beams of two antennas are facing in opposite directions, only the back lobe gains will be considered as aiding the interference. If the antennas are only partially facing each other, then only the partial gains are taken into account.

5.9 Other Sources of Signal Losses

Wireless signals are subject to numerous types of losses along the way, and the nature of the environment determines the kinds of losses encountered.

Scattering Scattering is a diffusion of signal energy as a result of impinging upon a rough surface. Instead of reflecting upon contacting a surface, instead the signal energy scatters in all directions. Examples of typical bodies in a mobile phone environment that cause scattering are trees, stones, and rough structures. To determine losses, one has to define the scattering coefficient.

Diffraction A phenomenon of a signal bending around a corner as it impinges upon a knife-edge or sharp object.

Reflection In trying to reach their destination, waves take an alternate route, especially when a LOS path is not available. As a result of reflection, a signal finds more than one path to move forward to the destination.

Penetration When a signal is completely blocked, it attempts to go through the obstacle, suffering significant loss. An example of such an obstruction is a building.

Shadowing This is a loss to a signal in space due to a profile change, such as terrain, buildings, trucks, or other cars.

Multipath Fading Fading suffered owing to destructive cancellation (partial or full) of two or more versions of the same signal arriving at the receiver via multiple paths.

Dispersion Loss encountered as a result of a lack of directionality of the propagation of the signal. This is based on an isotropic source radiating energy.

Path Clearance Loss Losses encountered as a result of not clearing an obstacle impinging on the Freznel zone. Typically, at least 60% of the 1st Freznel zone must be cleared for a point-to-point link.

Free Space Loss Losses dependent on distance and frequency. They are given by $PL_{fs} = 32.45 + 20\log_{10} d_{\text{km}} + 20\log_{10} f_{\text{MHz}}$.

Frequency Dispersion The relative motion between the transmitter and receiver causes the frequency of the carrier to be dispersed. This is called the Doppler spread.

Intersymbol Interference This is a loss caused by a signal arriving late at the receiver and impacting and causing interference with the next consecutive symbols that arrive ahead of it.

Intercarrier Interference Caused by a lack of time and frequency synchronization between receiver and transmitter (or a loss of orthogonality between subcarriers), this results in the skewing of the FFT window in the receiver.

5.10 Tower-Mounted Amplifiers (TMAs)

Used as means of balancing the uplink and downlink, TMAs extend the coverage of a cell site by lowering the receive sensitivity of the BS. TMAs are best used in suburban and rural areas.

Like anything else, TMAs have shortcomings. Namely, they are susceptible to interference. The fact that they make the BS sensitive also makes them susceptible to interference and intermodulation (IM) [7]. Hence, one must take this tradeoff into account when using TMAs. Figure 5.10 shows the mechanism involved in the installation and use of TMAs.

Let's quantitatively develop how TMAs draw their capability to lower the receiver sensitivity of a BS.

$$Sens_{RX} = N + NF + SNR \tag{5.13}$$

where N is total noise, $N = \mathrm{k}TBW$, k is the Boltzmann constant, T is the temperature in Kelvin, and BW is the bandwidth.

The noise figure is given by

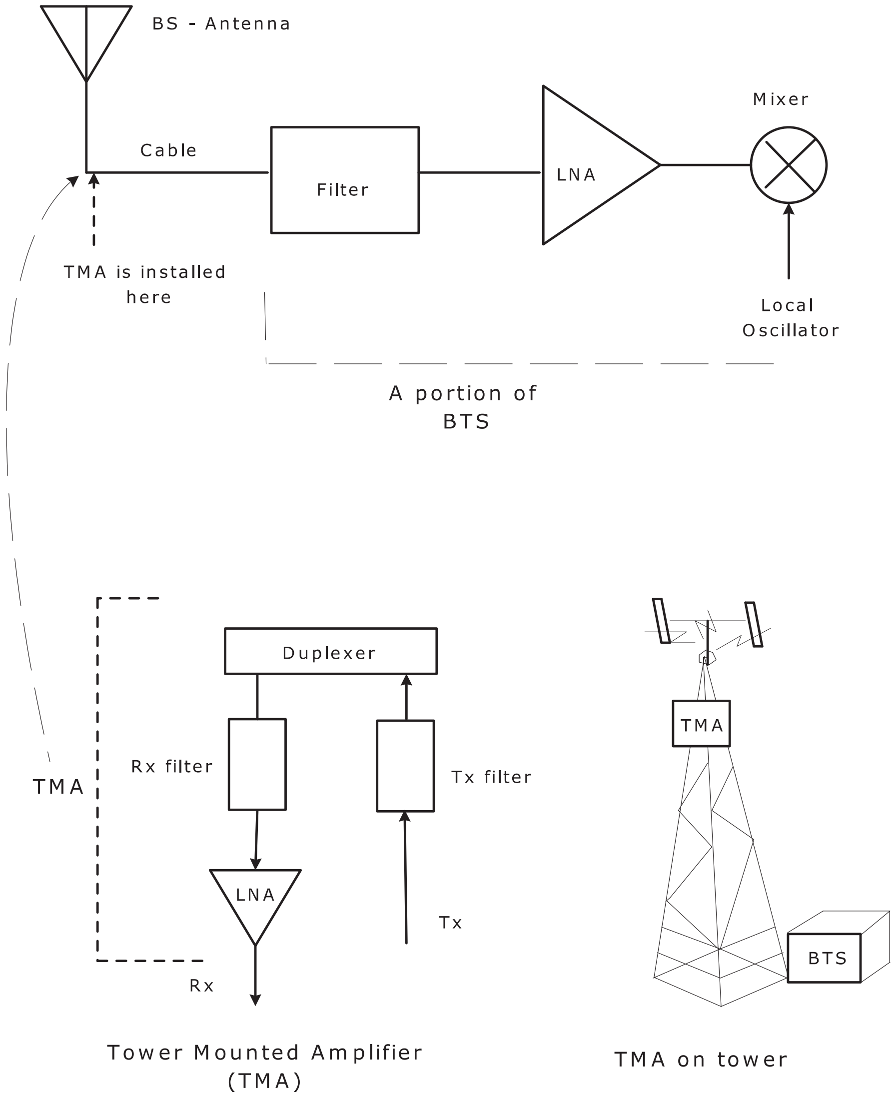


Figure 5.10 Tower-mounted antenna. (*After:* [7].)

$$NF = \frac{SNR_{IN}}{SNR_{OUT}} \tag{5.14}$$

NF is the noise figure, SNR_{IN} is the signal-to-noise ratio at the input of the receiver, and SNR_{OUT} is the signal-to-noise ratio at the output of the receiver.

The noise figure for the cascaded system is given by:

Table 5.1
TMA Gain Calculation

Reciever Data:						
N =	-118	dBm		Cable loss=	3	dB =2.0
SensRX =	-101	dBm		Cable gain=	0.5	
SNR =	7	dB	5.0			
SensRX =	NF+N+SNRimplying->		NF =	10		
NF(TOTAL)	=	2.0	+	18.0	=	13.0 dB
This will result in sensitivity of = NF (13) + SNR (7) + N (-118) = -98 dBm						
Receiver with TMA (adding TMA stage)						
TMA Data:						
NF	2.3	dB =	1.7	decimal		
Gain	10	dB =	10.0	decimal		
NF (TOTAL) =		TMA NF (dec)	2nd stage	3rd stage		
		1.7	0.1	1.8	=	3.6
Now the modified Receiver sensitivity is:				NF (3.6) + SNR (7) + N (-118) =		
				-107.4	dBm	
Improvement of over: 9 dB						

$$NF_T = NF_1 + \frac{NF_2 - 1}{G_1} + \frac{NF_3 - 1}{G_1 G_2} + \ldots + \frac{NF_i - 1}{G_1 G_2 G_{i-1}} \tag{5.15}$$

NF_T is the total noise figure, and G_i is gain of the ith stage.

From the NF_T, we can see that the first block, NF_1, is a limiting factor, especially if it is made of cable, and the loss associated with it. (Refer to Table 5.1.)

It can also be seen from the subsequent stages, that each stage is desensitized by the stage(s) previous to it, on account of the gains, $G_{i-1}, \ldots$, in the denominator. The second stage is an exception, since, looking at the second term on the right side of the equation, it cannot be desensitized, as the first stage does not have any gain. Instead G_1 happens to be a loss in this case. If you follow through the calculation in Table 5.1, you can see an improvement of 9 dB.

One important caveat that bears repeating in the use of TMAs is that they come with a cost; namely, they are susceptible to interference and IMs, since now the BS is rendered more sensitive to receiving weaker signals.

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Part III

WiMAX Radio Design

6

Network Planning Fundamentals

Before a system is designed, it is essential that an accurate link budget be developed and a quick feasibility study be done. The feasibility study determines the extent of the coverage, capacity, and cost of the project, among others.

These tasks can be handled with the use of a link budget, range calculation, flat-Earth coverage, and capacity determination spreadsheets. The next major and final step after this process, which is not part of this discussion, is design and deployment simulations with the help of a computer or software equipped with optimized model and terrain data.

The flat-Earth spreadsheet computation involving the link budget, range, and capacity determination steps should come reasonably close to the computer simulation step. Last but not least, the investment break-even point should be computed.

To make a good judgment, the designer should study the morphology (clutter category) and demography of the coverage area in detail and use his or her experience not only in interpreting the results, but also in adjusting the initial or final parameters to reflect the reality on the ground.

In the following, a typical link budget will be presented and discussed. Next, using a propagation model and the result from the link budget, a range will be computed. Finally, coverage, capacity, and cost of investment will be formulated and calculated.

6.1 WiMAX Link Budget

Refer to Figure 6.1 and the subsequent parameter definition in Table 6.1 to get familiarized with the link budget computation.

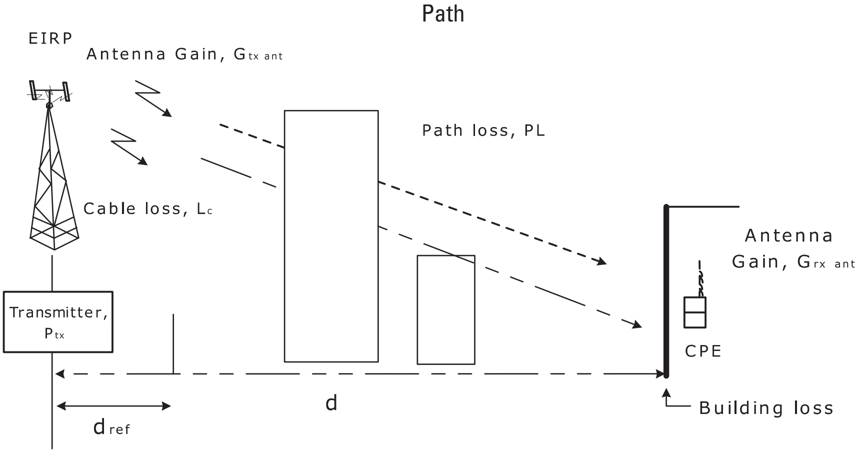

Figure 6.1 Link budget parameters.

Let's begin the link budget development for WiMAX by defining the key variable in Table 6.1.

Prior to developing a sample link budget, let's define key relationships that characterize transmitted power and received signal, with their respective antennas, as well as the channel between them.

Effective Isotropic Radiate Power, EIRP

$$EIRP = P_{TX} - L_C + G_{TX-Ant} \tag{6.1}$$

Receiver Sensitivity

$$S_{RX} = N + 10\log_{10} BW_{Hz} + NF + \frac{S}{N} - G_{SC} \tag{6.2}$$

Isotropic Receive Level

$$IRL = S_{RX} - G_{RX-Ant} \tag{6.3}$$

System Gain

$$G_{Sys} = EIRP - IRL_{Req} \tag{6.4}$$

Maximum Allowable Path Loss (Link Margin)

$$MAPL = G_{Sys} - M_{Tot} \tag{6.5}$$

Table 6.1
Link Budget Paremeters

Effective Isotropic Radiated Power	EIRP	Receive Antenna Gain	G_{RX-Ant}
Isotropic Receive Level	IRL	Maximum Allowable Path Loss	MAPL
Building penetration loss	L_{bldg}	Subchannelization Gain	G_{SC}
Fast Fading	M_f	Thermal Noise	N
Log-normal	M_{Logn}	Signal-to-Noise Ratio	$\frac{S}{N}$
Interference	M_{Int}	Noise Figure	NF
Total Margin	$M_{Tot} = L_{bldg} + M_f + M_{Logn} + M_{Int}$	System Gain	G_{Sys}
Cable Loss	L_c	Transmit Power	P_{TX}
Transmit Antenna Gain	G_{TX-Ant}	Receiver Sensitivity	S_{RX}
Free Space Path Loss	L_{fspl}	Transmitter	Tx
Bandwidth	BW	Receiver	Rx
Customer Premise Equipment	CPE		

The link budget shown in Table 6.2 is for outdoor mobile as well as indoor desktop environments. For each case, downlink and uplink are considered. The whole objective of the link budget (or that of any link budget) is to demonstrate how one typically uses the transmitter, receiver, and the various link margins and losses to deduce the maximum allowable path loss (MAPL). MAPL is a measure (and extent) of the reach of the link. In fact, right after the link budget, as will be shown later, MAPL will be used to calculate the radius of the cell. The larger the MAPL, the bigger the radius.

You can think of the link budget in the following way. The receiver has a sensitivity specification below which it cannot receiver an intelligible signal. The transmitter has limited power, even more limited in the case of a subscriber station (SS), due mainly to the life and size of its battery. On account of government regulations or intrasystem interference control, the media connecting the two (in this case, free space) is not as free as we would like it to be. For all intents and purposes, it is replete with losses related to obstructions, reflections, and diffractions, not to mention distance-related losses and, in the case of mobile SS, multipaths, and Doppler shifts.

You can then ask the following question: I have *X* dBm power that I can use to transmit, and if the receiver is a certain distance away, say, *D* km, and if,

Table 6.2
A Sample Link Budget

		Outdoor (Mobile)		**Indoor (Desktop)**	
Transmitter		**Downlink**	**Uplink**	**Downlink**	**Uplink**
1	Transmitter power output (dBm)	38	30	38	30
2	Cable loss, L_c (dB)	3	0	3	0
3	Antenna gain, $G_{tx\text{-}ant}$ (dBi)	17	0	17	5
4	Number of Tx antennas, (dec. to dB)	2	1	2	1
5	EIRP (dBm)	55	30	55	35
Receiver					
6	Antenna gain, $G_{tr\text{-}ant}$ (dB)	–1	18	6	18
7	Number of channels, Ch_{no}	16	16	16	16
8	Subchannel gain, G_{sc} (dB)	12	12	12	12
9	Noise figure, NF (dB)	8	5	8	5
10	Signal-to-noise ratio, SNR (dB)	1	2	1	2
11	Receive noise level, N (dBm/Hz)	–174	–174	–174	–174
12	Channel bandwidth, BW (dBHz)	10	10	10	10
13	Receiver sensitivity, S_{rx} (dBm)	–107	–109	–107	–109
14	Isotropic receive level, I_{RL} (dBm) (= $S_{rx} - G_{rx\text{-}ant}$)	–106	–127	–113	–127
15	System gain, G_{sys} (dB) (= EIRP – I_{RL})	161	157	168	162
Link Margins					
16	Log normal fading margin, Mln (dB)	8	8	8	8
17	Fast fading margin, M_{ff} (dB)	6	6	0	0
18	Interference margin, m_{int} (dB)	3	3	3	3
19	Building penetration loss, L_{bldg} (dB)	0	0	13	13
20	Total margin, M_{tot} (dB) (= $M_{ln} + M_{ff} + M_{int} + L_{bldg}$)	17	17	24	24
21	Max allowable path loss, M_{APL} (dB) (= $G_{sys} - M_{tot}$)	144	140	144	138

Note: It should be pointed out that, strictly speaking, for mobile applications, the number of carriers corresponding to PUSC is considered. In that case, the transmit power is given by per subcarrier, and the composite power is calculated by multiplying by the number of subcarriers used in that scheme. Likewise, on the receive side, sensitivity is given on a per-subcarrier basis and the total receiver sensitivity is given by multiplying by the number of subcarriers.

let's say, the receiver requires a Y dBm receiver sensitivity to be able to modulate the receive signal, how much do I have left over for losses and margins in order to close the link? Hence, margin = $X - Y$. Roughly speaking, is the meaning of a link budget. I can then use the margin just calculated, by applying a propagation algorithm, which will be shown later, to determine *D*. *D* is a measure of how far the receiver can be located away from the transmitter for this given margin: *D* = Function(MARGIN). It is strongly suggested that the reader study the link budget example given in Table 6.2 to draw enough appreciation.

6.2 Propagation Model and Range Calculation

For our purposes, the Stanford University interim (SUI) model has been chosen, not only as a range-determination tool, but also for our discussion of the propagation model algorithm and analysis. The choice is not necessarily because it is superior to others. For example, the Cost 231 model works just as well, provided it is adjusted for WiMAX frequency ranges. As a matter of fact, in numerous circles Cost 231 is preferred for its better prediction, barring its frequency-range limitation. However, the reader is free to choose the model he or she deems more appropriate for the application.

6.2.1 Prediction and Simulation

Due to the complexity of WiMAX, with OFDM, OFDMA, MIMO, and AMC propagation, prediction can be very difficult. The ray tracing model, for example, uses very fine-resolution terrain data, which can be very expensive. Numerous other prediction models used in prediction and simulations, largely on account of their frequency range limitations and to some extent because of accuracy issues, are deemed less fit. Simulation software requires good prediction algorithms, high-resolution terrain data, and a clutter model. Additionally, the algorithms need to be optimized for clutter correction, which is essentially a reality check.

It is advisable to use at least 30-m terrain data digitized from a 1:24k scale map. Propagation predictions are erroneous mainly because of clutter. Clutter, comprising foliage and man-made objects, are generally classified as:

- Dense urban;
- Urban;
- Suburban;
- Industrial;
- Agricultural;
- Rural.

Assigning a discrete value corresponding to a clutter type without taking the height of BS can result in wrong predictions. As a consequence, clutter model crudeness can result in ±6 dB (standard deviation) [1].

The Cost 231 model, also called the Hata model PCS extension, is extended to include the PCS frequency of 2 GHz. For the sake of completeness and comparison purposes, the Cost 231 path loss is given here:

$$L = 46.3 + 33.9\log\left(f_{MHz}\right) - 13.82\log\left(h_{te}\right) - C_H + \left[44.9 - 6.55\log\left(h_{te}\right)\right]\log\left(d_{km}\right) + C \tag{6.6}$$

where

f = frequency

d = distance

h_{te} = effective transmitter antenna height

C_H = SS antenna height correction factor, dB

Correction factor:

C = 0 dB for medium cities and suburban areas

C = 3 dB for metro areas

In order to use the Cost 231 model, the frequency factor needs to be adjusted.

6.3 Stanford University Interim Model (SUI Model)

SUI is selected here because:

1. It lends itself better to analytical steps.
2. It matches better with the power law propagation model.
3. Its parameters for terrain and the correction factors are defined in a straightforward manner.

Our sole purpose in this section is to take a link budget result and apply the model to compute a realistic range and get insight into the initial steps of the design process. These initial steps, incidentally, are necessary for decision making, budgeting, and proposal estimates. In the successive sections, further decision-making tools, such as coverage, capacity, and cost are outlined.

The SUI model is given by [2]:

$$L = A + 10\gamma \log_{10}\left(\frac{d}{d_o}\right) + X_f + X_h + S \tag{6.7}$$

where

L is the total loss, including margins;

A is a free space loss at the reference distance, d_o;

γ is the path loss exponent and is given by:

$$\gamma = \left(a - bh_b + \frac{c}{h_b}\right) \tag{6.8}$$

where

h_b is the height of the BS above ground ($10\text{m} \leq h_b \geq 80\text{m}$)

a, b, and c are constants, depending on the terrain type. Refer to Table 6.3.

where

d is the distance between BS and receiver ($d \geq d_o$);

d_o is 100m;

X_f is the frequency-correction factor and is given by:

$$X_f = 6 \log_{10}\left(\frac{f}{2{,}000}\right) \tag{6.9}$$

where

f is the frequency;

X_h is the correction factor for the height of CPE and is given by:

$$X_h = -10.8 \log_{10}\left(\frac{h_m}{2}\right) \text{ for terrain types a and b} \tag{6.10}$$

Table 6.3

SUI Terrain Definition

Constants	**Terrain A**	**Terrain B**	**Terrain C**
a	4.6	4.0	3.6
b	0.007	0.007	0.0065
c	12.6	17.1	20

$$X_h = -20 \log_{10}\left(\frac{h_m}{2}\right) \text{ for terrain type c} \qquad (6.11)$$

where

h_m is the CPE height above ground;

S is a shadowing factor, log-normally distributed random variable, typically between 8.2 and 10.6 dB.

In order to calculate the range (*d*) from MAPL in our link budget, we use the SUI model and solve it for *d*. The MAPL value in the link budget takes the place of L in the model. Please note that the range calculated is based on a flat Earth and does not take terrain variations into account. However, the clutter (or the morphology) parameters used and the margins taken into account will go a long way in approximating the reality on the ground.

Depending on multipath, LOS or NLOS, and the Doppler effect, the SUI model further defines two models for each type of terrain, resulting in a total of six terrain types.

A high multipath/delay implies a reflective environment, and a high value of K means more LOS, while a higher Doppler suggests a mobility environment. Refer to Table 6.4.

Now, in order to calculate the range for our link budget, here is the process:

1. Solve the SUI model equation for distance, *d*.
2. Replace *L* by the value of MAPL from the link budget.
3. Determine the terrain type the link budget is for.
4. Calculate X_f for the frequency.

Table 6.4

SUI Terrain and Channel

Model	Terrain Type	Delay	Rice (K)	Doppler Effect
SUI-1	c	Low	High	Low
SUI-2	c	Low	High	High
SUI-3	b	Low	Low	Low
SUI-4	b	Medium	Low	High
SUI-5	a	High	Low	Low
SUI-6	a	High	Low	High

5. Calculate X_h for the terrain type at hand.
6. Compute A, the free space path loss, based on the reference distance, d_o.
7. Using a, b, and c values corresponding to the terrain type and Xf and Xh, calculate gamma, γ.

Example:

From the link budget let's take MAPL = 138 dB, for an indoor uplink, since generally the uplink is the limiting link, on account of limited power of the SS.

Given the following assumptions:

$d_o = 100$ meters;

$S = 9$ dB;

CPE height, $h_m = 3$m;

Height of BS, $h_b = 20$m;

$f = 3.5$ GHz.

Assume terrain b.

$$L = MAPL = A + 10\gamma \log_{10}\left(\frac{d}{d_0}\right) + X_f + X_h + S$$

Solving for d,

$$d = \left\{ 10^{\left(\frac{MAPL - A - X_f - X_h - S}{10\gamma}\right)} \right\}(d_o) \tag{6.12}$$

Now we need to determine a, X_f, X_h, and γ.
a is the free space path loss for reference distance given by:

$$a = 20\log_{10}\left(d_{o(\mathrm{km})}\right) + 20\log_{10}\left(f_{\mathrm{MHz}}\right) + 32.45$$

$$a = 20 \times \log 10(0.1) + 20 \log 10\left(3.5 \times 10^3\right) + 32.45$$

$$= 83.3 \text{ dB}$$

Now from the height of CPE, h_m, 3m, we can calculate the correction factor for the CPE height.

$$X_h = -10.8 \times \log 10\left(3/2\right) = -1.9$$

From the frequency, *f*, we can compute, X_f

$$X_f = 6 \times \log 10\left(3.5 \times 10^3 / 2{,}000\right) = 1.5$$

Now the final variable to be calculated is γ:

$$\gamma = \left(a - bh_b + \frac{c}{h_b}\right)$$

Referring to Table 6.3 under Terrain B to substitute for *a*, *b*, and *c* and h_b, given as 20 meters

Then,

$$\gamma = \left\{4.0 - (0.007 \times 20) + 17.1\right\} = 4.75$$

Finally a figure for the range, *d*:

$$d = \left\{10^{\left(\frac{138-83-1.5+1.9-9}{10\times 4.75}\right)}\right\}(0.1) = 1.5 \text{ km}$$

This range or radius of a cell seems reasonable for a suburban environment.

6.4 Coverage and Capacity Planning

In this section, coverage and capacity will be computed.

Notice that the range computed in the previous section will be an input for the coverage computation. Coverage is dependent on the objective of the yearly percentages of areas, while the number of subscribers is dependent on the penetration levels for each year, pieces of information a marketing department normally provides. Refer to Table 6.5.

The result from this calculation, the number of coverage and capacity cell sites, will be used as input to compute breakeven in Section 6.5.

Table 6.5
Coverage and Capacity Spreadsheet

Coverage and Capacity Planning					
Coverage Area Definition	**Year 1**	**Year 2**	**Year 3**	**Year 4**	**Year 5**
Total area (km^2)	2,000	2,000	2,000	2,000	2,000
% area to be covered	15%	30%	60%	100%	100%
Area covered (km^2)	300	600	1,200	2,000	2,000
Penetration					
Morphology (clutter) specific number of households	412,000	424,360	437,091	450,204	463,710
Yearly growth	3%	3%	3%	3%	3%
Penetration rate	4%	6%	10%	12%	15%
Total subscribers	16,480	25,462	43,709	54,024	69,556
Data Rate, Oversubscription, Revenue/Month					
Maximum data rate/subscribers (Mbps)	2	2	2	2	2
Oversubscription	50	50	50	50	50
Revenue/subscribers/month ($/month)	$39	$38	$37	$36	$35
Band Usage					
Frequency band	3.5 GHz	(used for range computation)			
Channel bandwidth (MHz)	5				
Number of channels	2				
Bandwidth used (MHz)	10				
Coverage and Capacity Information					
Cell radius – (R, km)	1.25				
Cell area (= $2.6 \times R \times R$, km^2)	4.2				
BTS, downlink capacity (Mbps)	25				
Number of Coverage Sites					
Accruing number of sites	72	143	286	477	477
Traffic supported (based on 25 Mbps/BTS)	1,800.0	3,575.0	7,150.0	11,925.0	11,925.0
Number of subscribers	16,480	25,462	43,709	54,024	69,556
Downlink average data rate per subscriber (kbps)	109	140	163	220	171

Table 6.5 (continued)

Number of Capacity Sites					
Number of subscribers	16,480	25,462	43,709	54,024	69,556
Data rate/site (Mbps)	25	25	25	25	25
Oversubscription	50	50	50	50	50
Maximum data rate/subscriber (Mbps)—given	2	2	2	2	2
Number of subscribers/site = [(data rate/site) * (oversubscription)]/(data rate/subscriber)	625	625	625	625	625
Number of capacity sites = (number of subscribers)/(number of subscribers/site)	27	41	70	87	112

6.4.1 Capacity Considerations

If capacity sites exceed coverage sites, you can proceed in either of the following ways:

1. Take the maximum of the two. It is okay for a flat-Earth calculation. That is, take max{capacity sites, coverage sites}.
2. Interactively shrink the coverage radius until the coverage and capacity sites are about equal. This ensures that you are meeting the capacity requirement.

6.5 Investment and Break-Even Considerations

It is in the best interest of the service provider to understand the level and return on investment. To that end, one has to compute the number of months before the breakeven point, that is, the point at which the amount of initial and ongoing investments (capex and opex combined) matches the cumulative revenue.

The number of months before breakeven can be computed, given the following:

- Capital expense per site;
- Operating expense per site per month;
- Network-wide cost per site per month;
- Revenue per subscriber per month.

Then the number of months before break-even is given by:

$$\frac{\text{Number of capex sites} \times \text{cost/site}}{\left(\dfrac{\text{Revenue}}{\text{sub}} \times \text{number of subs}\right) - \left[\dfrac{\text{cost opex}}{\text{site}} + \dfrac{\text{cost system opex}}{\text{month}}\right]} \tag{6.13}$$

References

[1] Jacobsmeyer, J. M., "When Measurements Aren't Enough," *Mobile Radio Technology*, May 2008.

[2] Carniero, H., et al., "Software Planning Tool for WiMAX Networks," Technical University of Lisbon, Portugal, 2007.

7

Point-to-Point Link Design

7.1 Introduction

Point-to-point (PTP) microwave, in contrast to its counterpart, point-to-multipoint (PtMP), is a radio signal transmission from a single point to another single point, usually in frequency ranges well above 1 GHz and involving a LOS environment. Important to communications and communications networks, PTP microwave has gone through a few application evolutions. Initially PTP/microwaves were used to carry long distance and TV signals until overtaken by fiber optics. Today public safety radios use microwaves. Microwaves, owing to their reliability and cheaper cost, are still attractive options for many applications. LOS microwave have fewer signal impairment sources, namely, path loss and atmospheric conditions (such as rain). In WiMAX, PTP is used for interconnecting BSs, also known as *backhaul*. Refer to Figure 7.1, which shows PTP and PtMP in a typical WiMAX application.

While microwaves occupy wider bands on the order of 30 MHz and bit rates well in excess of 150 Mbps, they require an extremely high availability percentage in order to remain reliable. Typical availability percentages and bit error rates (BER) are on the order of greater than 99.995% and as low as (or lower than) 10^{-9}, respectively.

BSs in WiMAX networks are interconnected via LOS PTP microwave links. The frequency used for this link is in the 2- to 66-GHz range. To ensure LOS adherence, among others, a Fresnel zone has to be calculated.

Designing a PTP link for WiMAX entails characterizing and predicting impairments and calculating minimum transmitter and receiver heights, an RF link budget, and availability percentages.

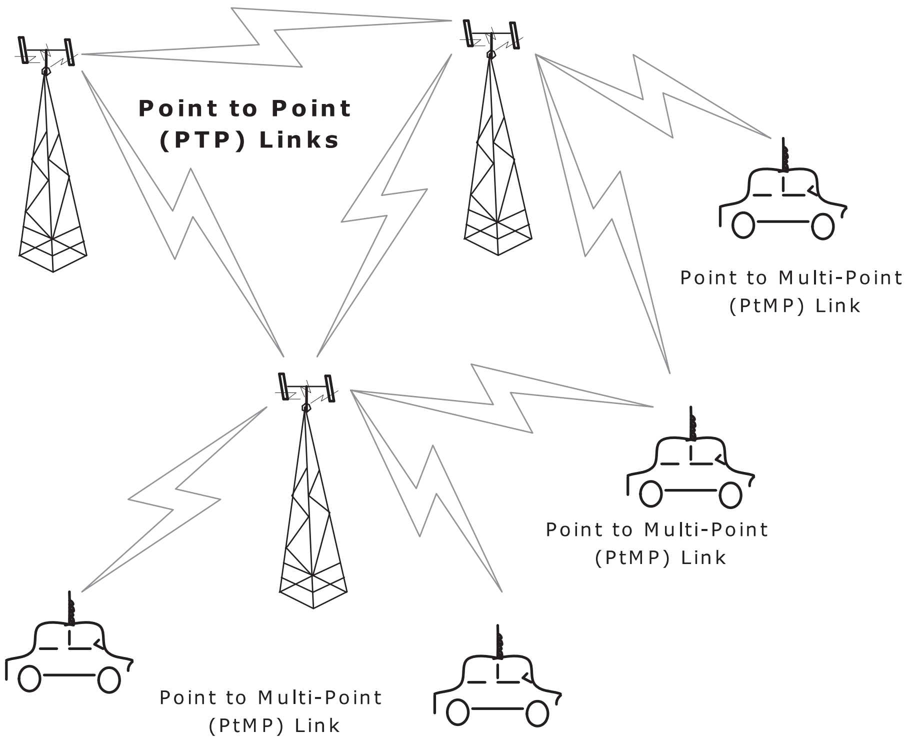

Figure 7.1 PTP and PtMP links in WiMAX.

In the following sections, the sources of impairments for PTP microwave will be characterized, followed by design examples.

7.2 Bandwidths, Rainfalls, and Margins

The need to use a higher frequency spectrum is motivated by a number of reasons, not the least of which is scarcity due to crowding at lower ranges, and availability of very large chunks of spectrum free of interference with capabilities to use smaller antennas. However, the losses incurred due to frequency and rainfalls are significant. That being said, spectrum is at a premium.

Speaking of the chunk of spectrum available at higher ranges, for example, if you take the 1-mm to 1-cm wavelength ranges, corresponding to 300 GHz to 30 GHz (also known as the microwave and millimeter-wave range), the difference in frequency between the two of 270 GHz is nine times more than all other bands combined. Table 7.1 is an example of frequencies versus wavelengths.

The first maxima of the atmospheric attenuation exists from 23 to 60 GHz. Rain leads to an increased scattering of waves, as the wavelength reaches

Table 7.1
Wavelengths for Important Frequencies

850 MHz	35 cm (cellular)
2 GHz	15 cm (PCS range)
WiMAX Ranges	
2.5 GHz	12 cm
3.5 GHz	8.6 cm
5.8 GHz	5.2 cm
10 GHz	3 cm
20 GHz	1.5 cm
Centimeter to Millimeter	
30 GHz	1 cm
40 GHz	0.75 cm
50 GHz	6 mm
60 GHz	5 mm
100 GHz	3 mm
200 GHz	1.5 mm
300 GHz	1 mm

the range of raindrop dimensions, resulting in further attenuation. Another major advantage of increased frequencies is the fact that, for a required antenna gain or beam width, the antenna area scales down with the square of the wavelengths.

For frequencies below 10 GHz, multipath is more dominant, and heavy rainfalls lower multipath fading. Hence, for frequencies above 10 GHz and heavy rainfall, multipath is lower. In this case, the allocated fade margin for the link can now be used for the rain attenuation.

Rain fading depends directly on frequency. It starts increasing at 10 GHz and increases very significantly after 15 GHz. Rain fading also depends directly on distance and is significantly pronounced above 10 km.

Vertical polarization antennas are less likely to be affected by rain attenuations.

Diversity, separating the paths by at least 8 km, results in diversity gain.

Heavy rainfalls lower multipath fading, which will preserve the fade margin for the rain attenuation.

Path availability, link reliability equal the percentage of time the received signal is above the required threshold. Availability is a function of, among others, the radio frequency, fade margin, path length, and local climate. If enough fade margin is built into the link budget, it makes the reliability better. For different parts of the world, depending on the climate, different fade margins are recommended. A typical design requirement is 99.995%, which is equal to an expected outage of 26 minutes per year [1].

A typical fade margin to account for fading and rain attenuation, usually is 25–40 dB. It is higher for more rain attenuation, larger distances, higher frequencies, and horizontally polarized antennas.

Figure 7.2 shows rain attenuation per kilometer versus frequency. There are four curves shown for each amount of rain: (1) drizzle, (2) light rain, (3) moderate rain, and (4) heavy rain. First the curve is identified for the type of rain and then the loss is read off the vertical axis corresponding to the frequency under consideration.

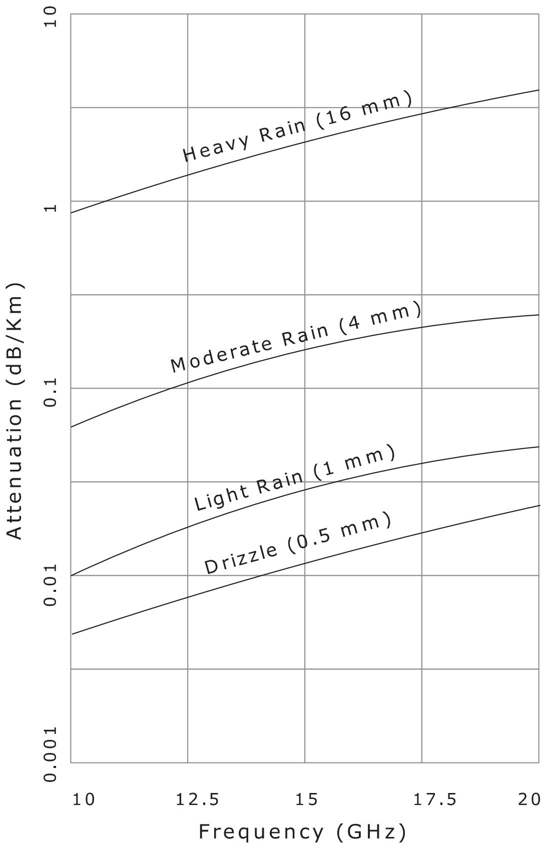

Figure 7.2 Rain attenuation corresponding to the type of rain and its frequency. (*From:* [2]. © 2005 Pendulum Instruments. Reprinted with permission.)

7.3 PTP Design Parameters and Signal Impairments

7.3.1 Refractivity

Electromagnetic waves traveling in space hardly travel in straight line. They bend up or down. Frequently, they bend or display such refractivity, in response to atmospheric conditions. This refractivity is more pronounced closer to the surface of the Earth.

$$\text{Effective Earth radius} = K \times \text{true Earth radius} \tag{7.1}$$

where true Earth radius = 6,371 km.

Then the *refractivity gradient* can be deduced from (7.1) by:

K = Effective Earth radius/true Earth radius

K compensates the refractive of the atmosphere.

Assumptions and Implications of K

The value of K captures the refractivity property:

- $K = 1$ implies no bending of electromagnetic wave.
- $K < 1$ implies bend up.
- $K > 1$ implies bend down.

A typical value used for K is 4/3.

A typical worst-case value for K is 2/3, requiring more transmitter or receiver height to avoid obstruction.

K = infinity is a theoretical value indicating an absolute flat Earth.

The practical implications are:

- Low K implies a more curved Earth. Terrain irregularity and man-made structure may intercept the Freznel zone.
- High K implies more flat Earth and better LOS (lower antenna height is obtained).
- $K < 1$ implies the signal is bent upward, away from the surface, requiring more height to avoid the signal's intercepting the Freznel zone.
- $K = 1$ implies no bending of the signal.

7.3.2 Foliage Attenuation

In addition to the atmospheric conditions that adversely affect signals, foliage losses can be serious sources of attenuations for PTP, or any signal for that matter. A remedy for this is to make a careful selection of the sites and then use a computer-aided link design with at least a 30-m terrain database. Make sure the database is current and that it reflects the presence of worst-case scenario foliage. Ideally, future changes in foliage depth should also be incorporated in the link design. Foliage attenuation can be calculated from

$$L = 0.2 f^{0.3} R^{0.6} \text{ in dB} \tag{7.2}$$

where

f is frequency in GHz.

R is depth of vegetation in meters (for $R < 400$ meters).

Diffraction Attenuation

Diffraction attenuation is incurred as a result of critical signal infringement on the first Freznel zone.

Rain Fading

Significant for long distance and especially for frequencies higher than 10 GHz, rain fading attenuates signals by absorption and scattering.

Rain drops smaller than λ absorb the wave energy, while raindrops close to a wavelength in size scatter. Rain fading also makes polarization discrimination difficult. In a rainy climate, it is advisable to use, if available, a lower frequency. Make sure you use a database containing the map relating the climate with the loss you have to budget into the design.

7.4 Availability

7.4.1 Path Availability

Path availability, also called link reliability, is the percentage of time a received signal strength is above a required threshold, S_{th}. The required threshold is given by:

$$S_{th} = -174 + 10 \cdot \log_{10}(BW) + NF + SNR + FM \tag{7.3}$$

where

BW = noise bandwidth in Hz;

NF = noise figure of receiver;

SNR = signal-to-noise ratio in dB;

FM = fade margin, to account for rain attenuation.

The required threshold can be broken down as follows:

$$ThermalNoise = -174\ \text{dBm/Hz}$$

$$TotalThermalNoise = -174 + 10.\log_{10}(BW)$$

$$SystemNoise = -174 + 10.\log_{10}(BW) + NF$$

$$ReceiverSensitivity = -174 + 10.\log_{10}(BW) + NF + SNR$$

Hence, S_{th} = receiver sensitivity + FM

$$Received\ Signal,\ S_{RX} = EIRP - PL_{fs} + G_{RXant} - L_{RXcable}$$

where PL_{fs} = free space path loss.

We compare S_{th} and S_{RX}, and the difference is a theoretical margin.

Path availability is a function of:

1. Frequency;
2. Diversity;
3. Fade margin;
4. Path length;
5. Local climate.

Availability is expressed as a percentage as:

$$\text{A} = 100 - \text{Outage (Unavailability)} \tag{7.4}$$

where Outage (Unavailability) (%) = SDS/Time · 100, and where SDS means significantly degraded second and time is the time period in seconds.

A digital link is deemed unavailable for service above and beyond 10 consecutive bit error rates (BER) of 10^{-3} SDS outage period.

7.5 PTP Design Examples

7.5.1 Design 1

Table 7.2 shows the design parameters and calculation corresponding to Figure 7.3's setup. The spreadsheet, in a straightforward manner, presents the calculations for the system operating margin, *RX* signal level, EIRP, and more importantly, the required signal threshold essential in computing the path availability of the PTP link. Important formulas are summarized at the bottom of Table 7.2 as well.

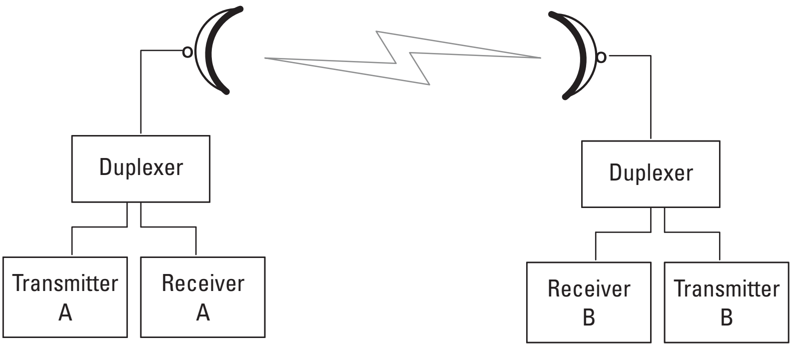

Figure 7.3 PTP microwave link.

7.5.2 Design 2

Refer to Figure 7.4.

Design 2 demonstrates the calculation of the height of the transmitter and receiver, taking into account the Freznel zone and refractive coefficient, *K*.

The radius of the Fresnel zone, *R*, is given by:

$$R = 72\sqrt{\frac{d_1 x d_2}{D_{tot} x f}}$$

A minimum of 60% of *R* must be clear.

Earth's curvature, *H*, is given by:

$$h = \frac{d_1 x d_2}{1.5 \times K}$$

where *K* is the refractive coefficient and a typical value for *K* is 1.33.

Then the minimum height of the transmitter, h_t, or receiver height, h_r, must be:

$$X + (R \times 0.6) + h$$

where *X* is the obstruction height, as shown in Figure 7.4.

Table 7.2

Design Parameters and Calculations

Design 1			
Received Signal Calculation			
Operating Frequency	2,400 MHz	**Link Separation**	30 miles
TX Antenna Gain	15 dBi	**RX Antenna Gain**	15 dBi
TX Cable Loss	3 dB	**RX Cable Loss**	3 dB
TX Power	30 dBm	**RX Sensitivity**	–95 dBm
Results			
Free Space Path Loss	133.696 dB		
EIRP	42 dBm		
RX Signal Level	–79.696 dBm		
System Margin	15.304 dB		
Required Signal Threshold Calculation		**Result**	
Bandwidth	30 MHz	**Required Signal Threshold**	–109.53 dBm
NF (Receiver)	15 dB		
SNR	20 dB		
Formulas:			
Free Space Path Loss = 92.467 + 20log10*(FGHz)* + 20log10(Dkm) *EIRP = TX Power – Ltx_cable + TX Antenna Gain* *Rx Signal Level = EIRP – PLfs + RX Antenna Gain – Lrx_cable* *System Margin = RX Signal Level - Receiver Sensitivity* *Required Signal Threshold,* S_{th} = –174+10log10*(BW) + NF + SNR + FM*			

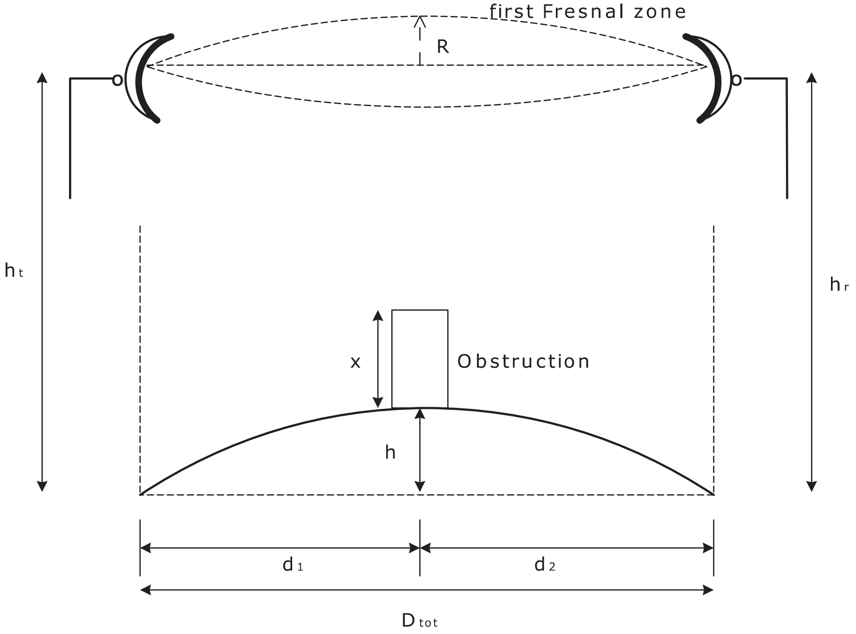

Figure 7.4 PTP microwave link, showing heights, taking the Fresnel zone and refractive gradient into account. (*From:* [2]. © 2005 Pendulum Instruments. Reprinted with permission.)

References

[1] Jacobsmeyer, J. M., "Designing Microwave Links," *Mobile Radio Technology*, April 2008.

[2] Duckworth, T., "Wireless Supporting Information," Pendulum Instruments, March 30, 2005.

8

Link, Design, Oversubscription, and Delay

In this chapter, two subtopics are presented. Section 8.1 covers digital link analysis and design, while Section 8.2 deals with service delays and customer oversubscription. In each case, adequate examples are given to illustrate the concept.

8.1 Link Analysis and Design

8.1.1 Digital Communication Link Design Example

To design a simple digital link, we need the following. First, we have to define $\frac{E_b}{N_o}$, the digital version of $\frac{S}{N}$.

$\frac{E_b}{N_o}$, the energy per bit-to-noise power spectral density, is a measure of the signal-to-noise ratio measured at the input of receiver. The level of $\frac{E_b}{N_o}$ determines the level of performance of the digital system.

$\frac{C}{N}$, the carrier power-to-noise power ratio, is defined as

$$\frac{C}{N} = \frac{E_b}{N_o} \cdot \frac{R}{BW} \tag{8.1}$$

where R is the data rate and BW is the bandwidth.

N, thermal noise, is given by:

$$N = kTB$$

where

k is the Boltzmann's constant, $1.380650 \times 10^{-23} \dfrac{J}{{}^{o}K}$

T is the effective temperature in Kelvin

B is the receiver bandwidth

The receiver's amplifier injects noise in the amplification process that is factored as noise figure, NF. Hence the total noise, system noise, or noise floor is then:

$$N_{floor} = N + NF \tag{8.2}$$

$\dfrac{C}{N}$ is now $\dfrac{C}{N_{floor}}$.

To get carrier power, C [1]:

$$C - N_{floor} = \frac{C}{N_{floor}} \text{ in dB}$$

$$C = \frac{C}{N_{floor}} + N_{floor} \tag{8.3}$$

Free space path loss is given by:

$$L_{fspl}(\mathrm{dB}) = 32.45 + 20\log_{10}(d_{\mathrm{km}}) + 20\log_{10}(f_{\mathrm{MHz}})$$

From $L_{fspl} = P_{TX} - P_{RX}$

Then,

$$P_{TX} = L_{fspl} + P_{RX}$$

Since P adding, fade margin, FM:

$$P_{TX} = L_{fspl} + P_{RX} + FM$$

Since P_{RX} is carrier power, C, then

$$P_{TX} = L_{fspl} + C + FM$$

Now, with a given value for E_b/N_o, distance, d, data rate, R, bandwidth, BW, and frequency, f, the transmit power required, P_{TX}, is calculated, and this is the power needed to close the link.

8.1.2 WiMAX Link Design Examples

Total path loss is given by:

$$L_{total} = 10.\alpha \log_{10}\left(\frac{d}{d_{ref}}\right) + L_{fspl} \tag{8.4}$$

where α is path loss exponent and L_{fspl} is free space path loss and is given by

$$L_{fspl}(dB) = 32.45 + 20\log_{10}\left(d_{ref}\right) + 20\log_{10}\left(f_{MHz}\right)$$

where

d = distance;

d_{ref} = reference distance (typically, the reference distance for macro is 1 km, for micro is 0.1 km, and for indoor is 1m);

f = frequency in MHz.

8.1.2.1 Example 1

Refer to Figure 8.1.

Assumptions:

- Frequency = 3.5 GHz;
- Transmitter power output, P_{TX}, = 30 dBm;
- BS coax loss = 2 dB;
- BS antenna gain = 9 dBi;
- Customer premise equipment (CPE) omni antenna gain = 3 dBi;
- Minimum bit error rate (BER) = 10^{-6}, corresponding to:
    - QPSK = –92 dBm;
    - 16 QAM = –84 dBm;
    - 64 QAM = –76 dBm.
- Building loss = 10 dB;
- Reference distance = 0.1 km;

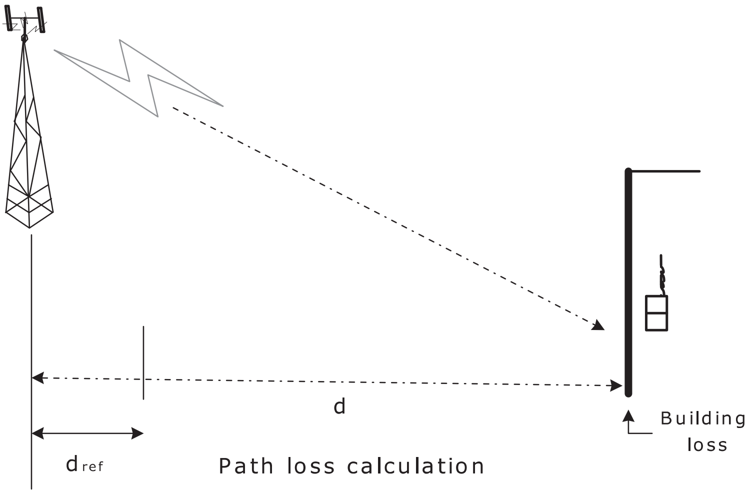

Figure 8.1 Link to inside of building and path loss.

- Path loss exponent = 3;
- Distance to CPE:
  - (a) $d = 2$ km;
  - (b) $d = 600$m;
  - (c) $d = 200$m.

8.1.2.2 Determine Which Distance Can Be Served with Which Modulation Type

For d = 2 km

Using total path loss: $L_{total} = 10 \cdot \alpha \log_{10}\left(\frac{d}{d_{ref}}\right) + L_{fspl}$, where

$$L_{fspl}(dB) = 32.45 + 20\log_{10}\left(d_{ref}\right) + 20\log_{10}\left(f_{MHz}\right)$$

$= 30 \times 3 \times \log_{10}$ (2 km/0.1 km)

$+ \{20 \log_{10}$ (0.1 km) + 20 $\log_{10}$ (2 × 103) + 32.45}

(The terms in { } are the free space path loss quantities.)

= 79.7 dBm

39 + {−20 + 66 + 32.45}

= 117.5 dB

EIRP = transmitter power output (P_{TX}) + antenna gain − coax loss

= 30 dBm + 9 dBi – 2 dB

= 37 dBm

Signal after traversing the distance and through the building wall:

= EIRP – L_{total} – L_{bldg}

= 37 dBm – 117.5 dB – 10 dB

= –90.5 dBm

We can see that the signal is enough to support QPSK level modulation.

For d = 600 km

EIRP is the same, 37 dBm;

L_{bldg} is the same, 10 dB;

L_{total} = 106.7 dB;

Signal through the building wall

= EIRP – L_{total} – L_{bldg}

= 37 dBm – 106.7 dB – 10 dB

= –79.7 dBm

The adaptive modulation and coding scheme in WiMAX can serve this distance with 16 QAM (which requires –84 dBm).

For d = 200m

L_{total} is the only thing that changes. As a matter of fact, free space path loss remains the same throughout this calculation.

L_{total} = 97.5 dB

Signal through the building wall is

= EIRP – L_{total} – L_{bldg}

= 37 dBm – 97.5 dB – 10 dB

= –70.5 dBm

The adaptive modulation and coding scheme in WiMAX can serve this distance with 64 QAM (which requires –76 dBm).

8.1.2.3 Example 2

Given the following receiver sensitivities at 10^{-6} BER:

- –85 dBm for BPSK;
- –82 dBm for QPSK;
- –76 dBm for 16 QAM.

and also given are the receiver antenna gain, 22 dBi, and EIRP, 37 dBm.

Calculate

1. The maximum allowable path loss;
2. Carrier-to-noise ratio in the above receiver sensitivities.

The receiver antenna gain and receiver sensitivity are given. To calculate maximum allowable path loss (MAPL), proceed as follows:

The isotropic receive level (IRL), similar to EIRP except that IRL is for the receiver side, is given by: IRL = receiver sensitivity − receiver antenna gain.

1. receiver sensitivities: −82/−85/−76 dBm

   IRL = −85 dBm − 22 dBi
   = −107 dBm
   MAPL = EIRP − IRL
   = 37 dBm − (−107) dBm
   = −144 dB

   MAPL (for receiver sensitivity = −82 dBm)
   = 37 dBm − (−104) dBm = 141 dB
   −141 dB

   MAPL (for receiver sensitivity = −76 dBm)
   = 37 dBm − (−98) dBm
   = −135 dB

2. Find *C*/*N* for each receiver sensitivity (−85/−82/−76 dBm) given *BW* = 10 MHz, *NF* = 9 dB. Thermal noise is given by:

   $N = kTB = -174$ dBm/Hz + 10 log 10^7
   −174 dBm/Hz + 70 dB(hertz) = −104 dBm
   (for 10 MHz *BW*, total noise)

   Noise floor is given by:

   $N_{floor} - N + NF$
   = −104 dBm + 9 dB
   = −95 dBm
   C/*N* (for −85-dBm receiver sensitivity)
   $C - N_{floor}$ (in dB), subtract noise floor from receiver sensitivity
   = −85 dBm − (−95) dBm
   =10 dB

C/N (for -82-dBm receiver sensitivity)
$= -82 \text{ dBm} - (-95) \text{ dBm}$
$\underline{= 13 \text{ dB}}$

C/N (for −76-dBm receiver sensitivity)
$= -76 \text{ dBm} - (-95) \text{ dBm}$
$\underline{= 19 \text{ dB}}$

8.1.3 Cochannel Interference and C/I Computation Example

Please refer to Figure 8.2.

Let distances to sites A, B, and C be d_A, d_B, and d_C, respectively

Let's also assume the target site is A and the other sites, B and C, are interfering sites.

The interference from sites B and C, attenuated by the path loss and building loss, are therefore I_i where (i = B, C).

I_B is given as $I_B = EIRP_B - L_{totalB} - L_{bldg}$.

L_{totalB} is the total path loss between site B and the building, as defined in Example 1. Similarly, interference from C is given as $I_C = EIRP_C - L_{totalC} - L_{bldg}$.

Therefore generalizing, the total interference is given by:

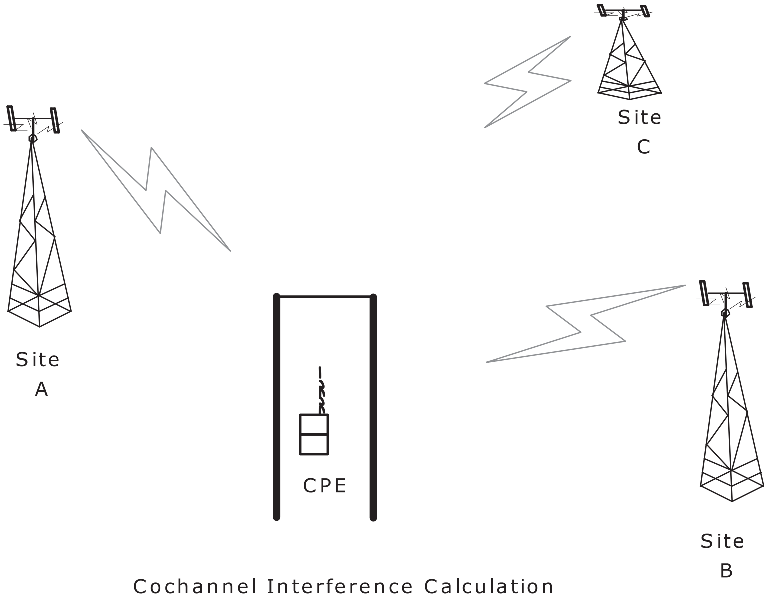

Figure 8.2 Cochannel interferences from two sites.

$$I_{Total} = \sum_{i=B}^{C} I_i$$

C to *I* is therefore:

$$\frac{C}{I_{Total}} = \frac{EIRP_A - L_{totalA} - L_{bldg}}{\sum_{i=B}^{C} I_i} \text{ dB} \tag{8.5}$$

8.2 Oversubscription, Link Usage, and Delay

Service oversubscription relies on the fact that not all subscribers access the system all at once—trunked. This allows the network provider to overcommit on the capacity of the system. Table 8.1 shows typical subscription ratios for various services.

This overcommitting is probably similar to what the airlines do. They keep statistics of no-show passengers, and based on this figure, they raise the number of tickets sold proportionally. Occasionally they can get in trouble, when and if they get their statistics wrong or overplay their hand. Wireless service providers do the same thing by allowing customers to sign up for services over and above the oversubscription ratio for which the system is designed.

8.2.1 Link Usage

Link usage is a ratio of the actual data rate, based on actual generated traffic, to the bit rate capability of the sector. If, for example, the traffic generated on a

Table 8.1
Oversubscription Ratio

Service Category	Oversubscription Ratio
Web surfing	10:1 to 25:1
VoIP	5:1 to 10:1
Multicast/unicast, video/ audio services	1:1
Videoconferencing	1:1 to 2:1
Internet gaming	5:1 to 10:1

Source: [2].

sector is 2 Mbps (on the uplink) and if the sector capability bit rate is 4 Mbps, the link usage is 50% (2/4).

Now, let's say a Web surfer is on a WiMAX system, and the oversubscription ratio is 15:1. For this simple example, the sector can be rated, taking into account oversubscription, 4 Mbps · 15 = 60 Mbps.

As the system traffic generated increases the system delay, the latency a subscriber experiences increases. A Web surfer waits more and more time for a browser to fill up a Web page. This takes us to the important concept of *probability of delay.* Since each subscriber's experience is different, delay can only be described by statistics.

Probability of delay is the concept of quantifying the likelihood of a delay in getting service.

8.2.2 Delay

At the outset, let's recall that WiMAX, like any other IP network, works based on the blocked call delayed or Erlang C concept. This means any subscriber asking for service, upon getting blocked (for insufficiency of capacity, for example), will never be turned down. Rather, the user will be kept in a queue almost indefinitely. This is unlike a cellular voice network that works on the concept of blocked call cleared or Erlang B, in which a subscriber will be dismissed right away.

In the following sections, we will discuss Erlang C, since WiMAX is based on it. For the sake of completeness and to shed important contrasting light on the differences, an Erlang B discussion and example are also included. Erlang B has the flavor of the trunking concept, because it calculates blocking probability rather than delay probability.

We will begin our discussion with the Poisson arrival process, the foundation for both Erlang B and Erlang C. We will provide examples along the way to make the concepts clearer.

8.2.3 Poisson Arrival Process

Customers arriving into a bank or at a grocery checkout counter do so in randomly spaced time intervals. For example, during a particular minute, 3 customers arrive at the checkout counter to form a line; in the next minute only 1 customer shows up, and a total number of 0 customers show up in the next 5 minutes. The average arrival rate would then be (3 + 1 + 0 + 0 + 0 + 0 + 0)/7 = 0.57 customer/minute. In IP, application packets arrive at a router, destination, or buffer, at randomly spaced times also. Customers requesting a wireless service—be it landline phone, cellular phone, or Web access—all follow the Poisson arrival process.

The following is a concrete example using the Poisson model. The Poisson arrival process is given by:

$$P_n(t) = \frac{(\lambda t)^n e^{-\lambda t}}{n!} \tag{8.6}$$

where

t = the time interval between 0 and t

n = the total number of arrivals in the interval 0 to t

λ = the total average arrival rate in arrivals/second

Now let's assume $n = 0$ (a simpler case), meaning a total of 0 arrivals between 0 and t. Hence, (8.6) simplifies to the following:

$$P_0(t) = e^{-\lambda t} \tag{8.7}$$

For example, let's take two cases in which: (1) $\lambda = 1/10$ and (2) $\lambda = 2/10$ (that is, 1 arrival and 2 arrivals in 10 seconds, respectively). The probability of seeing no arrival in the interval 0 to t is plotted in Figure 8.3.

It can be seen that, as the average arrival rate, λ, is decreased ($\lambda = 0.1$), the probability that there will be 0 arrivals increases. For a larger number of seconds, the probability of 0 arrivals decreases exponentially, in either case.

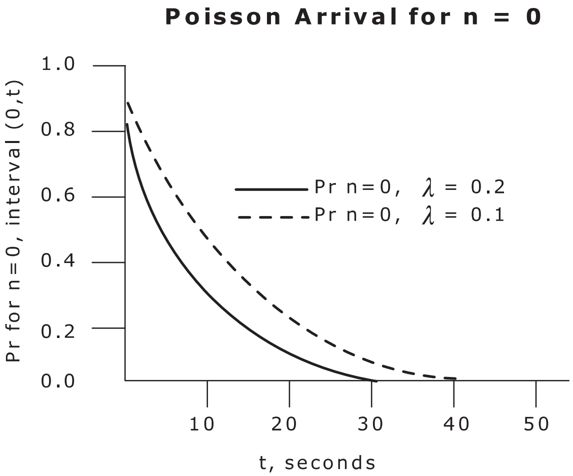

Figure 8.3 Poisson arrival.

8.2.4 Erlang B

A Danish mathematician, Agner Krarup Erlang, invented the Erlang model, as well as traffic engineering and queuing theory. The unit Erlang measures traffic intensity in telecommunications systems.

If a channel in a trunked system is occupied for 1 hour (60 minutes) that constitutes 1 Erlang. If 10 users were to make a call and each call duration were 4 minutes, then a total of 40 minutes would be used, and the hours of traffic would be 40/60 = 0.67 hour or 0.67 Erlang. The Erlang model also enables one to calculate blocking probability, a measure of grade of service (GOS). The Erlang B model assumes callers encountering busy signals are blocked and (unless we are using the Extended Erlang B model) will not retry. The Erlang B model is usually presented in table format with three variables, busy hour traffic in Erlang, GOS (blocking probability), and the number of lines (channels) in a trunk group. If you know any two of the variables, you can find the third one.

In Erlang B, blocking probability, or probability of *k* active users, is given by:

$$P_k = \frac{\dfrac{\left(\dfrac{\lambda}{\mu}\right)^k}{k!}}{\left(\displaystyle\sum_{j=0}^{m} \frac{\left(\dfrac{\lambda}{\mu}\right)^j}{j!}\right)} \tag{8.8}$$

where = λ, as defined before, is a Poisson process arrival rate, and $\frac{1}{\mu}$ is service times or call duration. Hence, $\frac{\lambda}{\mu}$, dominant in (8.8), can be thought of as a measure of traffic intensity, or offered traffic with a unit of Erlang. First we determine $\frac{\lambda}{\mu}$. We can determine $\frac{\lambda}{\mu}$ from the following $E[K]$, the mean number of active users, given by [3] as

$$E[K] = \frac{\lambda}{\mu}(1 - P_m) \tag{8.9}$$

where $\frac{\lambda}{\mu}$, as in (8.8), is offered traffic in Erlang. P_m is the probability of blocking. Intuitively, $1-P_m$ is the percentage of unblocked calls. Then $E[K]$ is the fraction of users requesting service and granted service. Let's assume the parameters in Table 8.2.

In Table 8.2, $E[K]$ is given 1,000, 1,500, and 1,750 call-seconds for BS1, BS2, and BS3, respectively. Let's calculate the offered traffic in each case. Thus, $\frac{\lambda}{\mu}$ for:

- BS1: $1{,}000 = \frac{\lambda}{\mu}(1-0.25)$, $\frac{\lambda}{\mu} = 750$, implying $750/3{,}600 = 0.208$ Erlang.
- BS2: $1{,}500 = \frac{\lambda}{\mu}(1-0.35)$, $\frac{\lambda}{\mu} = 975$, implying $975/3{,}600 = 0.271$ Erlang.
- BS3: $1{,}750 = \frac{\lambda}{\mu}(1-0.3)$, $\frac{\lambda}{\mu} = 1{,}225$, implying $1{,}225/3{,}600 = 0.340$ Erlang.

We add all of the Erlangs, since all are in a trunk group. Thus, the total is 0.819 Erlang.

Now, to compute the probability of blocking, or that all of the three servers are blocked, from (8.8).

Table 8.2
Erlang Calculation Example Values

Base Stations	**Call-Seconds in Busy Hour (3,600 in One Hour)**	**Blocked Calls (%)**
BS1	1,000	25
BS2	1,500	35
BS3	1,750	30

$$P_3 = \frac{\dfrac{\left(\dfrac{\lambda}{\mu}\right)^k}{k!} = \dfrac{(0.819)^3}{3!}}{\left(\begin{array}{l} \displaystyle\sum_{j=0}^{3} \dfrac{\left(\dfrac{\lambda}{\mu}\right)^j}{j!} = \dfrac{(0.819)^0}{0!} + \dfrac{(0.819)^1}{1!} + \\ \qquad\qquad\qquad \dfrac{(0.819)^2}{2!} + \dfrac{(0.819)^3}{3!} \end{array}\right)} = 4.1\% \tag{8.10}$$

8.2.5 Erlang C (Blocked Calls Queued)

In Erlang B, users are blocked when a channel is unavailable. However, in Erlang C, users are kept in a queue. (Technically, calls are kept in the queue for a certain queuing time T_Q,[1] imposed by the switch.) Hence, predicting the probability of delay in the queue for unavailable service is defined by Erlang C model, usually given in the form of a table. The probability of a queue forming, or that an arriving call will encounter a busy system, is given by the following relationship:

$$\Pr(delay > 0) = \frac{A^C}{A^C + C!\left(1 - \dfrac{A}{C}\right)\displaystyle\sum_{k=0}^{C-1} \dfrac{A^k}{k!}} \tag{8.11}$$

where A is total traffic offered (in Erlangs) and C is the total number of resources available.

The condition under which Erlang C works is the following:

- Service requests are Poisson arrival.
- Service times are exponentially distributed.
- Service requests from customers are independent from each other.
- Resource allocated exclusively to one customer is for the specified period (they can be shared).
- Number of customers is much larger than number of resources available by at least a factor of 10.

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[1] Time in the system, T_Q = waiting time, T_w + service time, T_s.

Also called *blocked calls delayed*, in Erlang C blocked calls are held in a queue and delayed until a channel is available.

Equation (8.11) describes only the probability that a call will not be granted service and will experience delay. But we need another equation to describe the probability a delayed call waits, say, more than t seconds.

$$\Pr\left(T_Q > t\right) = \Pr\left[\text{delay} > t \middle| \text{delay}\right] = \exp\left(-(C - A)t/H\right) \tag{8.12}$$

This equation answers the following question: Once a call is delayed, what is the probability it will be delayed more than t seconds?

And yet another problem needing characterization is the following: What is the probability that, among all calls, including those serviced immediately, a call is delayed more than t seconds? The answer is given by

$$\Pr\left[\text{delay} > t\right] = \Pr\left[\text{delay} > 0\right] \times \Pr\left[\text{delay} > t \middle| \text{delay}\right] \tag{8.13}$$

This seeks to find the intersection of delayed calls and those delayed more than t seconds, and multiplies the two probabilities given in (8.11) and (8.12).

To find GOS (delayed call), first find the probability that a call is initially denied access to the system. This is determined by (8.11). Second, for delayed calls forced to wait t seconds, use (8.12). Next, to find the probability that any call has to wait more than t seconds, use (8.13). Hence, the probability (a call is delayed) · (conditional probability) (delay > t second | given a call is delayed).

This is also illustrated in Figure 8.4 and an example.

Hence, the GOS of a trunked system in which blocked calls are delayed is given by

$$\begin{aligned}&\Pr\left(\text{delay} > t\right) = \Pr\left(\text{delay} > 0\right) \cdot \Pr\left(\text{delay} > t \middle| \text{delay} > 0\right)\\&= \text{pr}\left(\text{delay} > 0\right) * \exp\left(-(C - A)t/H\right)\end{aligned}$$

where C is the number of trunked channels, A is the total offered traffic, and H is an average duration of a call.[2]

For derivation of the previous, refer to [4].

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[2] $A_u = \mu H$, where A_μ is traffic intensity by each user, u is the average number of calls per unit time, and H is the average duration of the call. $A = UA_\mu$, where A is the total traffic intensity and U is the total number of users.

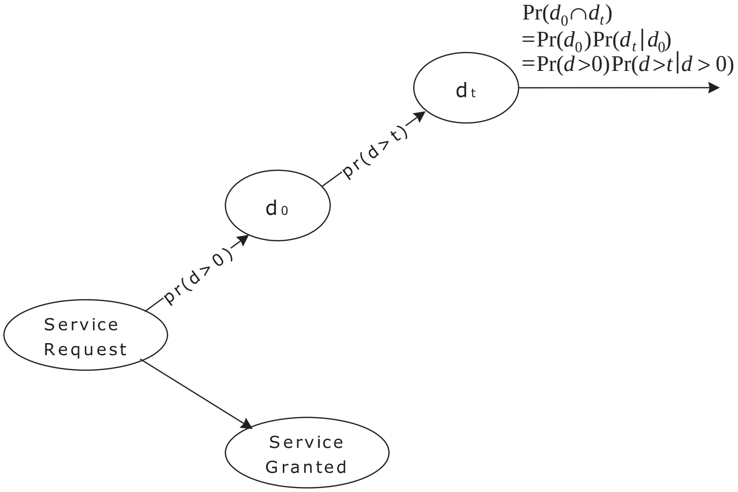

Figure 8.4 Erlang C probability computation—a call delayed > *t* seconds.

8.2.5.1 Erlang C Example

A trunked channel, $C = 20$, GOS (= prob(delay > 0) is given by 4% and load per user, A_μ, is given by 0.022 Erlang per subscriber. The number of call request, μ, is given as two calls per hour.

1. Find the probability a delayed call will have to wait for more than 15 seconds.
2. Find the probability a call is delayed for more than 15 seconds.

a. Given $\mu = 2$ calls/hour, then the holding time would be: $H = 0.022/2 = 0.011$/hour = 39.6 seconds. From the Erlang C (Appendix B) table corresponding to $C = 20$ channels, GOS = 0.05, and $A = 13$ Erlangs, hence

$$\Pr(\text{delay} > t | \text{delay} > 0) = \exp(-(C-A)t/H)$$

$$= \exp(-(20-13)15/39.6)$$

$$= 7\%$$

b. GOS = 5% = 0.05

$$\Pr(\text{delay} > 15) = \Pr(\text{delay} > 0) \cdot \Pr(\text{delay} > 15 | \text{delay} > 0)$$

$$= 0.05 \cdot 0.07$$

$$= 0.35\%$$

The answer of 0.38% seems credible, since the number of trunked channels is high and the GOS is relatively low.

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[1] Pearce, J., "What's All This E_b/N_o Stuff Anyway?" *Spread Spectrum Scene Online*, 2000.

[2] Bellavich, R., and R. Olexa, "WiMAX Forum RF Network Engineer Certification Boot Camp," *DoceoTech*, 2008.

[3] Jacobsmeyer, J. M., "The Ins and Outs of Traffic Engineering," *Mobile Radio Technology,* August 2006.

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9

WiMAX Capacity, Frequency Planning, and MIMO Antennas

In this chapter, three important and distinct, yet indirectly related, topics are covered:

1. WiMAX capacity;
2. Frequency planning;
3. Multiple-input-multiple-output (MIMO) antennas.

The WiMAX capacity subsection demonstrates the strength of WiMAX technology to deliver at a high data rate. Section 9.2 shows that, through deliberate arrangement and assignments of frequencies to a group of cells, interference is minimized, thereby optimizing coverage and capacity. MIMO, likewise, by making use of multiple antennas at either end of the wireless communication, transmit and receive, improves SNR.

9.1 WiMAX Capacity

A WiMAX capacity computation quantitatively shows why and how tightly and closely spaced subcarriers result in high data rates. Figure 9.1 shows subcarriers in frequency and time domains.

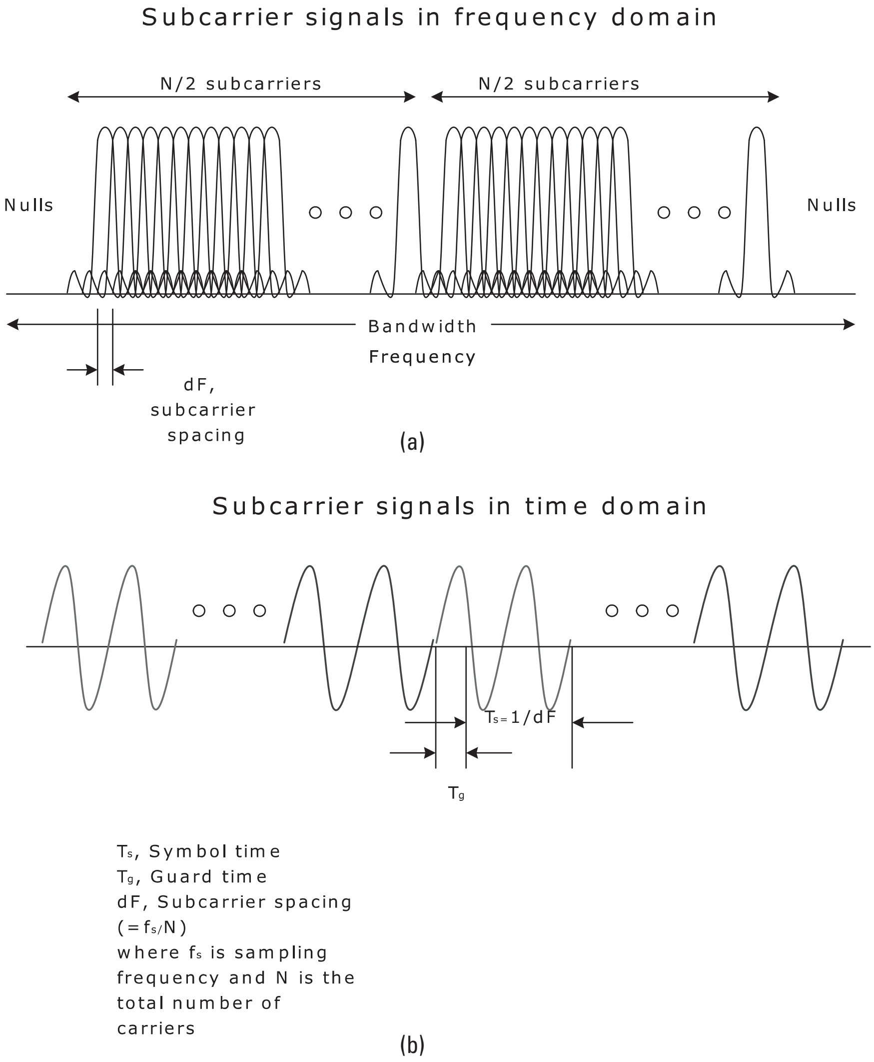


Figure 9.1 Subcarriers in (a) frequency and (b) time domains.

9.1.1 Capacity Calculation

Roughly speaking, the trick in capacity calculation is the identification of the total number of bits to be transmitted and the time available (including overhead) to do so, thus taking the ratio of the former to the later. In order to arrive at a generalized capacity derivation, we begin by defining each variable that comprises the final formula. Please refer to Figure 9.1. It is instructive to point out the N subcarriers shown in Figure 9.1 are transmitted at the *same time*.

$$f_s = floor\left(nBW/8{,}000\right)8{,}000 \tag{9.1}$$

where f_s is the sampling spectrum, and n is the sampling factor. This parameter depends on the bandwidth, as defined in Table 9.1.

BW is the nominal channel bandwidth in Hz. B_{sub} is the spacing of the sampling spectrum or subcarrier bandwidth. It is given by

$$B_{sub} = \frac{f_s}{N_{tot}} \tag{9.2}$$

where N_{tot} is the total number of subcarriers, and N_{data} is the number of data subcarriers.

T_s is the symbol time, the inverse of which is B_{sub},

$$T_s = \frac{1}{B_{sub}} \tag{9.3}$$

Appending time, symbol time plus guard time, T_{s+g}, is given by:

$$T_{s+g} = T_s\left(1 + F_g\right) \tag{9.4}$$

where F_g is the guard factor and is a fraction of the form,

$$F_g = \frac{1}{2^a} \text{ where } a = \{2, 3, 4, 5\} \tag{9.5}$$

$$T_g = T_s \,.\, F_g \text{ where } T_g \text{ is guard time} \tag{9.6}$$

$$T_s = \frac{1}{dF} \text{ where } dF \text{ is subcarrier spacing } \left(= fs/N\right) \tag{9.7}$$

Table 9.1
Sampling Factor

If *BW* Is a Multiple of	Then *n* (Sampling Factor) Is
1.25	144/125
1.5	86/75
1.75	8/7

Capacity calculation in frequency domain is given as follows:

$$R_{data} = \frac{B_{sub}.N_{data}.Mod\_level.conv\_coderate.block\_coderate}{1+F_g} \tag{9.8}$$

Data rate, *R*, in bits per second =

[(Bsub (=sampling spectrum /total number of FFT used, unit:
Hz = 1/second) ·
Ndata (no. of data carriers = number of data symbols) ·
Mod_levels, modulation levels (unit: no. of bits/symbol) ·
Conv_coderate, coding rate (represents the reduction of the
convolution coding, unit: constant)]
Block_coderate (unit: constant) /
1+F_g (factoring in guard overhead, unit: constant)

Mod_level, modulation levels, (in bits/symbol) is given by
M is the number of possible states, from base conversion,

$$\log_2 M = 3.32 \log_{10} M \tag{9.9}$$

(for $M = 16$ corresponding to 16 QAM, 4 bits/symbol is transmitted).

It is instructive to keep in mind that all the symbols are transmitted at the same time in their allotted spectrum. That is what gives WiMAX its high data rate.

conv_coderate is convolution code rate, which uses the usable number of bits. Typical code rates are 1/2, 2/3, 3/4. For example, in case of 1/2, for every bit there is one additional redundant bit used.

block_coderate is Reed-Solomon block coding that can further reduce the usable number of bits. When 0.85 rate is used, for example, it implies that 15% is overhead.

Capacity data rate in time domain:

$$R_{data} = \frac{N_{data} \cdot Mod\_levels \cdot conv\_coderate \cdot block\_coderate}{T_{s+g}} \tag{9.10}$$

9.1.1.1 Example

Calculate capacity using frequency domain and time domain methods for the following data.

$BW = 10$ MHz, $N_{data} = 768$, $N_{tot} = 1{,}024$, $M = 16$, conv_coderate = 3/4, block_coderate = 0.90,

$F_g = 1/8 = 0.125$, $n = 8/7$.

The rest of the terms are calculated as follows:

Then $f_s = 11{,}424$ Hz

$B_{sub} = f_s/N_{tot} = 11{,}156.25$

$T_s = 89.64\ \mu s$

$T_s + g = T_s(1 + 0.125) = 100.84\ \mu s$

Mod_levels = Log2(M) = 4

Then in time domain:

$$R = (11{,}156.25 \cdot 768 \cdot 4 \cdot 0.75 \cdot 0.9)/1.125 = 20.6 \text{ Mbps}$$

In frequency domain:

$$R = (768 \cdot 4 \cdot 0.75 \cdot 0.9)/100.84 = 20.6 \text{ Mbps}$$

A good figure of merit to find out if this data rate is reasonable or to quickly compare capacity with other channel sizes is the spectral efficiency.

Spectrum efficiency = data rate/BW given by bps/Hz

For our example, the spectrum efficiency would be 20.6 Mbps/10 MHz = 2.1 bps/Hz (more on the level of WiFi).

The conditions assumed in the example suggest an adverse environment. A more favorable outcome would have been closer to 5 bps/Hz. However, comparing the result to Table 9.2, it closely matches the downlink rate corresponding to 16 QAM, 3/4 code rate. Judging from the rates, the downlink/uplink ratio appears to be about 6:4.

9.2 WiMAX Frequency Planning

Frequency planning is the centerpiece of network planning success in WiMAX. As such, there is ongoing research to come up with better and improved techniques to minimize potential or existing interferences and meliorate spectrum efficiency.

Normally, the available bandwidth, especially in scalable OFDMA or PUSC, is divided into subchannels. A subset of these subchannels can then be

Table 9.2
Mobile WiMAX Data Rates

Mod.	**Code Rate**	**5-MHz Channel**		**10-MHz Channel**	
		Downlink Rate, Mbps	**Uplink Rate, Mbps**	**Downlink Rate, Mbps**	**Uplink Rate, Mbps**
QPSK	1/2 CTC, 6x	0.53	0.38	1.06	0.78
	1/2 CTC, 4x	0.79	0.57	1.58	1.18
	1/2 CTC, 2x	1.58	1.14	3.17	2.35
	1/2 CTC, 1x	3.17	2.28	6.34	4.70
	3/4 CTC	4.75	3.43	9.50	7.06
16 QAM	1/2 CTC	6.34	4.75	12.67	9.41
	3/4 CTC	9.50	6.85	19.01	14.11
64 QAM	1/2 CTC	9.50	6.85	19.01	14.11
	2/3 CTC	12.67	9.14	25.34	18.82
	3/4 CTC	14.6	10.28	28.51	21.17
	5/6 CTC	15.84	11.42	31.68	23.52

Source: [1].

assigned to various sectors of a BS. WiMAX utilizes both licensed, as well as unlicensed, spectra. Channel bandwidths are multiples of 1.25 MHz, 1.5 MHz, and 1.75 MHz, with a maximum of 20-MHz bandwidth. WiMAX supports two modes of operations: time-division duplex (TDD) and frequency-division duplex (FDD).

FDD divides the available spectrum exactly in half, for downlink and uplink. TDD uses the same spectrum for downlink and uplink, separated only in time. TDD is asymmetrical and more amenable to applications, such as the Internet. The uplink and downlink percentage in TDD is decided according to the data rate requirements, with a ratio of 1:1 being equal for downlink and uplink. For example, 1:2, 1:3, and 1:4 ratios signify that 2 times, 3 times and 4 times more frames, respectively, are assigned to the downlink than the uplink. The service provider can elect which ratio to use, according to the capacity and revenue objectives. In any event, these facts are transparent for our frequency-planning task at hand. The frequency assignments apply, irrespective of the ratio used in TDD or whether an FDD scheme is used.

9.2.1 Fractional Frequency Reuse

Fractional frequency reuse is a frequency-planning technique that optimizes throughput and spectrum efficiency [2]. One of the critical aspects or a potential interference source is frame synchronization between BSs using the same channels or a different UL/DL ratio in TDD for each BS. Both of these scenarios create BS interference. Like a cellular system, WiMAX similarly assigns to each sector the channels with a certain frequency reuse factor across the service area. What is unique about WiMAX is that it has a built-in mechanism that tries to minimize subcarrier collisions from interfering cells using the same channels. That is where permutation (PUSC and FUSC subchannelization schemes) comes in. PUSC and FUSC subchannelization schemes randomly form slots from the available subcarriers based on a predetermined permutation (each BS can be assigned a different permutation.) This reduces the chances that interfering cells are using the same subcarrier, even when they are operating on the same channel. However, when the system is loaded, the reduction in interference achieved through diversity subchannelization methods rapidly becomes less effective.

A fractional frequency reuse scheme is given below. In this frequency-planning scheme (please see Figure 9.2) the available bandwidth is divided into three

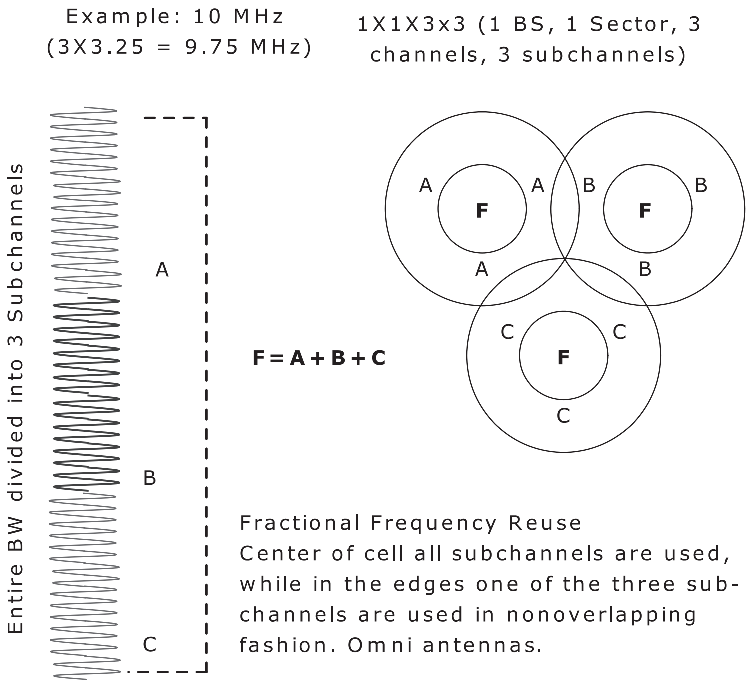


Figure 9.2 Frequency planning: 1 × 1 × 3 × 3.

subchannels, and all the subchannels will be available to subscribers closer to the BS (maximizing spectrum efficiency). A subset or, most specifically in this case, one-third of the subchannels will be available in the outlying areas. Overlapping sectors use a different subset of the subchannels, effectively preventing interference.

9.2.2 Frequency Plan (Other Schemes)

Proposed frequency plans to consider are shown in Figures 9.3–9.6. While these configurations are not meant to be for specific situations, they will no doubt help, at a minimum, navigate and expedite a design process. In nearly all cases, the attempt is to improve spectrum efficiency, increase throughput, and avoid potential interferences.

9.3 MIMO Antennas

9.3.1 Overview

As a compelling and up-and-coming technology, WiMAX makes use of the latest advances in antenna designs.

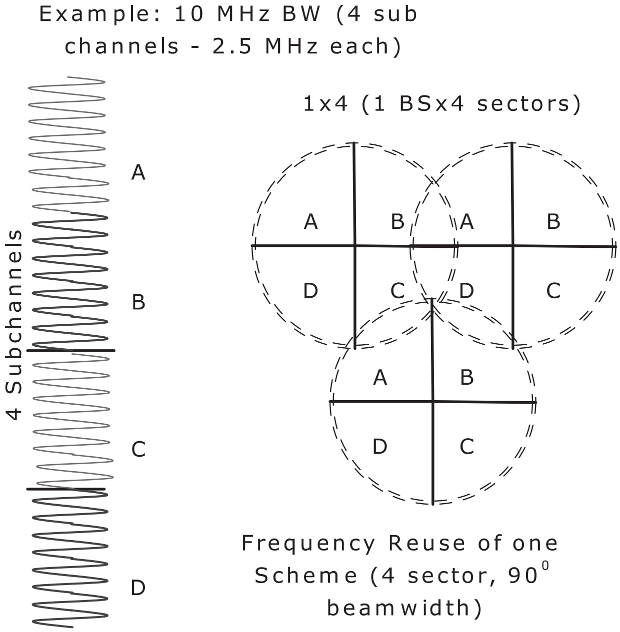


Figure 9.3 Frequency planning: 1 × 4 × 1 × 4.

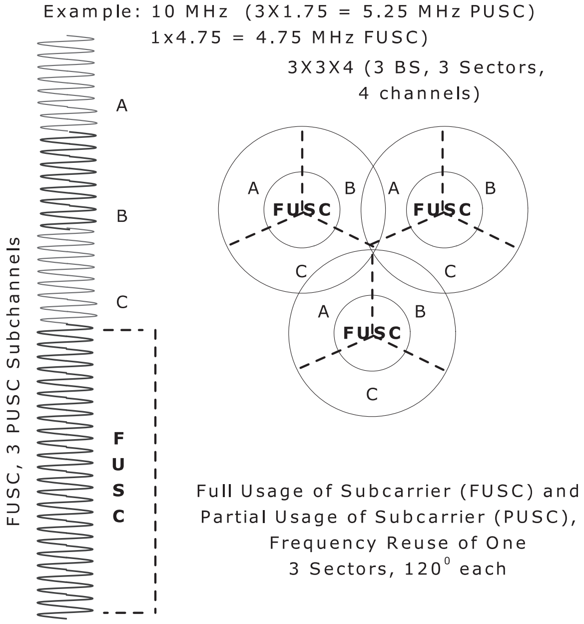


Figure 9.4 Frequency planning: 1 × 3 × 1 × 4.

As pointed out previously, the main objective of a communication network is to enhance coverage and capacity, while minimizing interferences in its various forms and eliminating or filtering out the effect of noise.

Meeting capacity demand by increasing spectrum is not the luxury most carriers have on account of its scarcity and cost.

Antenna diversity, especially receive diversity techniques— based on putting two antennas sufficiently apart—have been around for a while and have proved very useful in shoring up faded received signals or improving SNR. Signals fade because of a variety of reasons, not the least of which is interference.

Traditionally [3], we lower transmit power, tilt, or use sectored antennas to alleviate interference and improve signal-to-noise + interference ratio. Unfortunately, more needs to be done. Multipath propagations, on account of reflections or alternate paths along the way, unless mitigated, can create interference challenges.

With the advent of multiple-input-multiple-output (MIMO) antennas, an additional degree of freedom became available as a tool to improve SNR. Single-input-and-multiple-output (SIMO) and multiple-input-single-output (MISO) antennas are the subsets of MIMO. MIMOs (much like CDMA uses a rake receiver to collect multipath signals) turn around the adverse effects of multipaths, with the help of digital signal processing (DSP), to improve SNR. As a matter of fact, the very idea of using multiple antennas, in both transmit

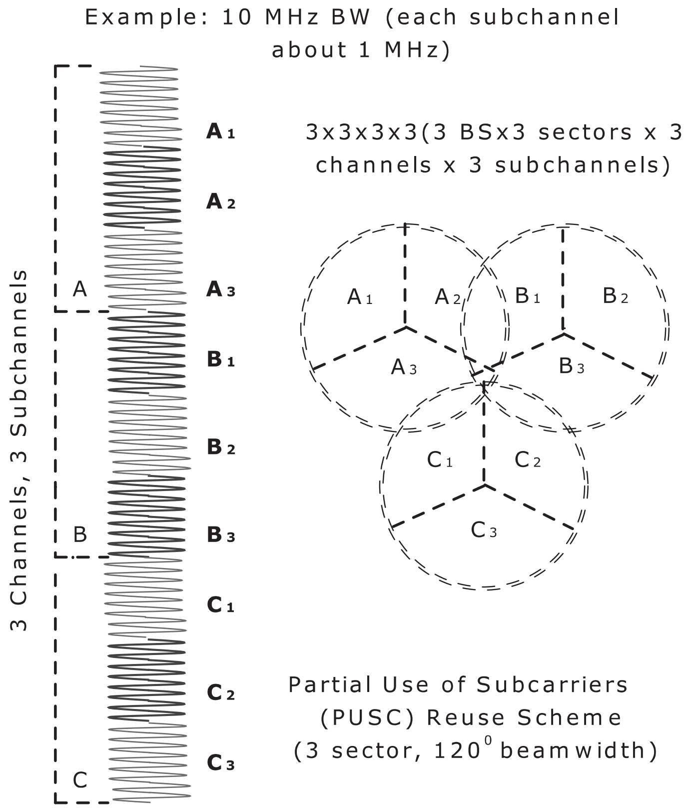


Figure 9.5 Frequency planning: 3 × 3 × 3 × 3.

and receive sides, to intentionally create additional multipaths is used as a backdoor to improve SNR.

9.3.2 Terms and Concepts

Spatial Diversity Antenna diversity can be viewed as a redundancy in the spatial domain and implemented by using multiple antennas at both the transmit side (base station) and the receive side (mobile units).

Transmit Diversity While it doesn't improve the SNR, on account of a power penalty for transmitting redundantly, however, it is a good technique to maintain signal levels (SNR) in the face of fading.

Receive Diversity SNR grows linearly with number of antennas, while capacity grows logarithmically. Installing diversity antennas on the BS is especially beneficial, if the signal on the uplink can be improved while SS is transmitting on a

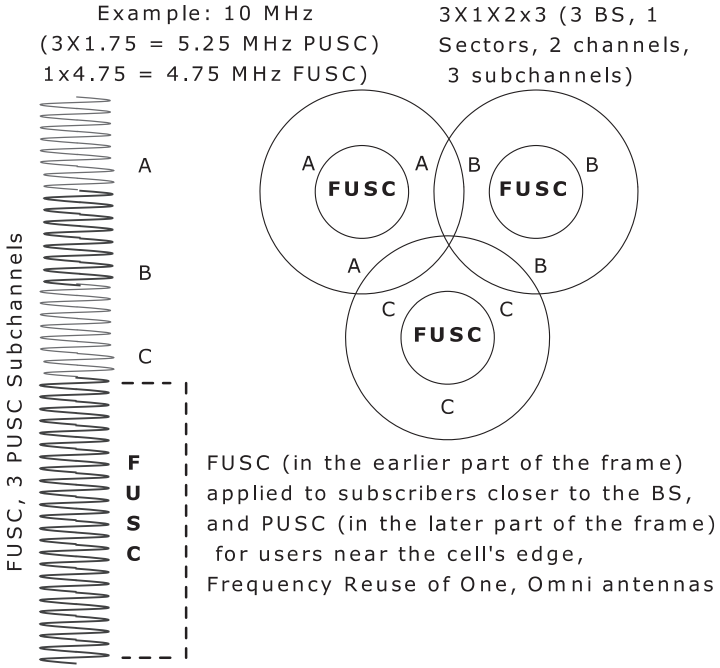


Figure 9.6 Frequency planning: 3 × 1 × 2 × 3.

single antenna. Multiple antennas at the mobile station are not as convenient and efficient if at all.

Closed Loop Closed loop is a MIMO configuration or situation, except in the case of time division duplex (TDD), where the knowledge of the channel is fed back to the transmitter.

Temporal Diversity Temporal diversity is a case in which channel coding is combined with time interleaving to give rise to redundancy in the time domain.

Frequency Diversity Frequency diversity is a case in which the same data is transmitted in more than one frequency, effectively creating redundancy. If frequency fades in one then more than likely it does not on another one.

Space-Time Code The coding technique that adds redundancy spatially as well as temporally, for multiple-antenna transmissions, is called space-time coding. Space-time code (STC) is the technique of transmitting multiple and redundant signals using multiple transmit antennas to improve the reliability of the receive signal, with the notion that not all transmission paths will be equally affected adversely by the channel.

This technique achieves both transmit diversity and coding gain. When coupled with multiple receive antennas, it can achieve the capacity of MIMO. There are numerous variations of STC, namely, coherent (the receiver having knowledge of the channel through training), noncoherent (neither the transmitter nor the receiver knows the channel), differential (amenable for slowly fading channels), and space-time trellis or block code (using different types of codes.)

Let's point out that 802.11n also uses MIMO technique and delivers 100–140 Mbps in a 40-MHz bandwidth, as opposed to 802.11g/a 54 Mbps in a 20-MHz bandwidth. As mentioned above, MIMOs turn around the challenges brought about by multipaths and use them to advantage.

9.3.3 Open Loop MIMO

9.3.3.1 MIMO, MATRIX A—STBC (Space-Time Block Coding)

This technique generates multiple versions of a datastream (a block of data at a time), encodes them into a space-time block code and transmits them across a number of antennas simultaneously.

Each transmitted signal is orthogonal and, hence, avoids self-interference at the receiver. The receiver can then either choose the strongest signal or use a maximal ratio combiner (MRC) to combine them. This scheme is used to enhance coverage.

Space-time block code implies, in addition to the diversity created by using multiple transmit antennas, the space part, using temporal diversity includes channel coding in conjunction with time interleaving, creating diversity in the time domain.

According to Shannon information theory, there are two ways to increase the capacity (data rate) of a communication channel:

1. Increase transmission bandwidth;
2. Increase transmission power (or decrease noise/interference) in SNR.

In STBC, as a result of generating redundant information, the error performance is improved, which implies you can use the one of the higher order modulation constellation (QPSK, 16 QAM, 64 QAM, and so on) to increase the data rate. This implies that, using this scheme, we are able to increase (trade error performance with data rate) data rate, without either increasing transmission bandwidth or transmission power.

9.3.3.2 MIMO, MATRIX B—SM (Spatial Multiplexing)

In this technique, the data stream is split into multiple streams and transmitted from multiple antennas (but within the same time time-frequency resources). Because of multipath, the various streams arrive at the receiver at sufficiently different spatial signatures, whence the distinction is made. This approach is for increasing capacity.

Adaptive mode (AM) uses either matrix A or matrix B, as described earlier, depending on the environment. In an environment in which SNR is low, matrix A performs better, while in areas in which SNR is high and bandwidth is limited, matrix B performs more optimally. In a WiMAX employing adaptive mode, matrix A, and matrix B, the system must compute and decide which one of the two matrixes works better.

Similar to adaptive modulation and coding (AMC), which decides and applies the correct modulation scheme based on the SNR, AM works the same way.

9.3.4 Beam Forming

There is yet another smart-antenna technique that a WiMAX system can take advantage of, namely, beam forming. Sections 9.3.4.1 and 9.3.4.2 describe two types of beam-forming techniques.

9.3.4.1 Direction-of-Arrival and Angle-of-Arrival Beam Forming

In the direction-of-arrival (DOA) and angle-of-arrival (AOA) methods of beam forming, the receiver's signal processing assigns a weight to each received signal and forms a beam in favor of the target signal, while suppressing the beam toward the interferer.

9.3.4.2 Eigenbeam Forming

Another type of beam forming, eigenbeam forming, computes the channel's impulse response of each receive antenna element, with an eye toward maximizing SNR. Using this knowledge, the signal will be transmitted by focusing it to a desired user.

9.3.5 MIMO Antenna Concept

Refer to Figure 9.7, which shows a MIMO antenna configuration.

Each receive antenna on the right is configured to receive a signal plus its reflected version from all the transmit antennas on the right. In the figure, for the sake of clarity, only two transmit antennas and one receive antenna are shown in action.

The channel, as discussed throughout this book, represents numerous unfavorable environments, among them being multipath and noise. The noise is AWGN (zero mean) and can be easily filtered out. However, multipaths will be used to our advantage, as described next.

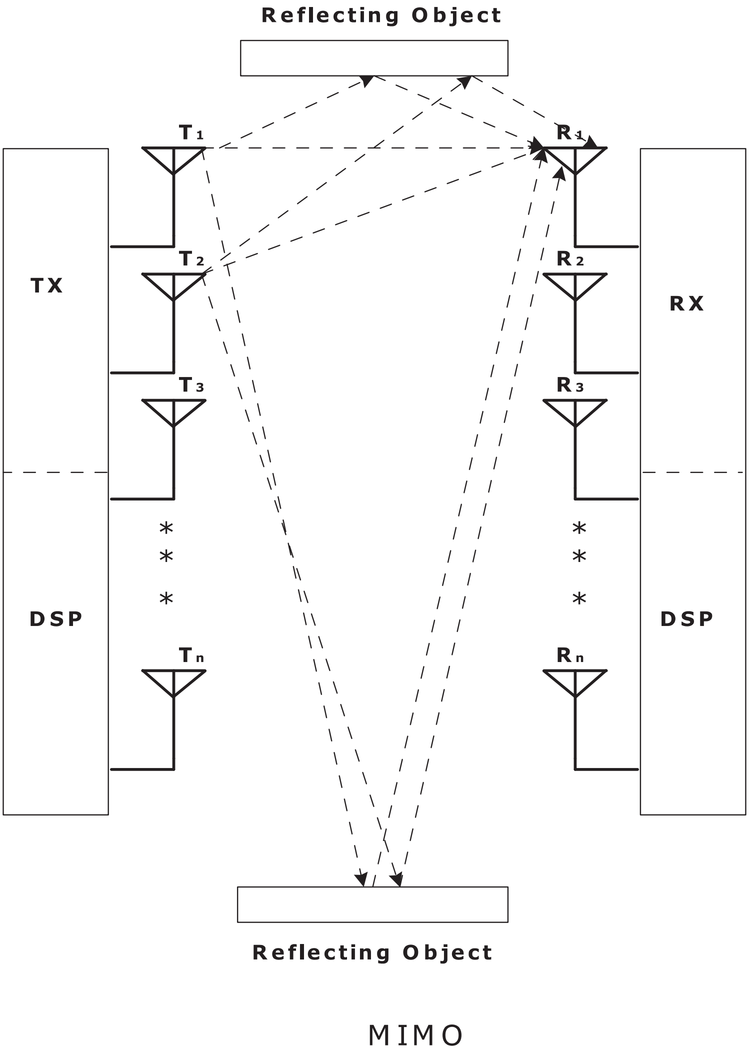


Figure 9.7 MIMO antenna concept.

The general form of receive signal is given by: $R = hT$. That is, the receive signal equals the channel (h) multiplied by the transmitted signal (T). Receive equations (9.11) and (9.12) apply to n transmit and n receive antennas.

R_i is the ith receiver, T_i is the ith transmitter and h_{jk} is channel connecting the jth receiver with the kth transmitter. Here are all the combinations:

$$
\begin{aligned}
R_1 &= h_{11}T_1 + h_{12}T_2 + \dots h_{1n}T_n \\
R_2 &= h_{21}T_1 + h_{22}T_2 + \dots h_{2n}T_n \\
&* \\
&* \\
&* \\
&* \\
R_n &= h_{n1}T_1 + h_{n2}T_2 + \dots h_{nn}T_n
\end{aligned} \tag{9.11}
$$

The matrix form of the above group of equations is:

$$[R] = [H][T] \tag{9.12}$$

In order to determine the received signal, the receiver must first estimate the channel, H, by piecing together information from within the packet more particularly, looking at the preamble part of the packet and applying signal processing. Then the transmitted signal, T, is calculated through

$$[T] = [H]^{-1}[R] \tag{9.13}$$

where $[H]^{-1}$ is the inverse of the matrix H.

There are n equations and n variables. However, there is additional information from the multipaths, which will make the solutions to the equation simpler. In other words, the reflected signal, say, for example, T_1, has two other versions (reflections) of itself: T_{1r_1} and T_{1r_2}. Hence, the receive signal containing h_{11} has three terms as opposed to just one.

$$R_1 = h_{11}T_1 + h_{11}T_{1r_1} + h_{11}T_{1r_2} \tag{9.14}$$

So, in this case, there are three terms for every transmitter, T_i. This will effectively increase the number of equations over and above the number of variables by 3:1 and, consequently, make the equation easier to solve for the unknown channel.

The foregoing MIMO discussion introduces only the fundamental concept and mathematical foundation. For a detailed treatment of the topic, the reader is advised to refer to other antenna textbooks.

9.3.6 Summary of Major Antenna Techniques

9.3.6.1 Spatial

1. Single-transmit and more than one receiver antennas (traditional antenna configuration)—Received signals at multiple antennas, sufficiently separated, result in diversity gain and improved SNR.
2. STC uses multiple transmit antennas and a single receive antenna. The same data coded differently and sent via multiple antennas results in both diversity and coding gain.

9.3.6.2 MIMO

1. Combines MISO and SIMO, hence, there is diversity gain.
2. MIMO improves capacity (throughput), hence resulting in both diversity gain and improved capacity.

9.3.6.3 Adaptive Antenna System

The *adaptive antenna system* (AAS) (also called adaptive beamforming system) maximizes SINR by creating multiple beams that target individual remotes. The main lobes are directed to the strongest signal and the side lobes to multipath signals, while the nulls are directed to interfering signals.

References

[1] WiMAX Forum, "Mobile WiMAX Part 1: A Technical Overview and Performance Evaluation," February 2006.

[2] Upase, B., M. Hunukumbure, and S. Vadgama, "Radio Network Dimensioning and Planning for WiMAX Networks," *Fujitsu Sci. Tech.*, 2007.

[3] Mukherjee, S., "WiMAX Antennas Primer—A Guide to MIMO and Beamforming," Motorola Networks and Enterprise, May 2007, http://www.wimax.com.

10

Case Study

The following discussion is an example of a preliminary radio-planning exercise. By taking into account coverage area, demographics, service provider investment objective, available spectrum, and design parameters, this chapter demonstrates how to determine the preliminary number of coverage and capacity sites in preparation for the finalized design.

A finalized radio frequency (RF) design typically involves running a computer simulation on a well-optimized propagation model and drive tests as an accuracy verification and correction step. Provided good engineering common sense is used, the preliminary design thus obtained should be very close to the final design.

As the final step of this case study, the break-even point for investment is also formulated.

10.1 Range Determination

Refer to Figure 10.1, which shows service area and clutter categories for the case study.

The morphology/clutter categories are:

- Dense urban (DU);
- Urban (U);
- Suburban (SU);
- Rural (R).

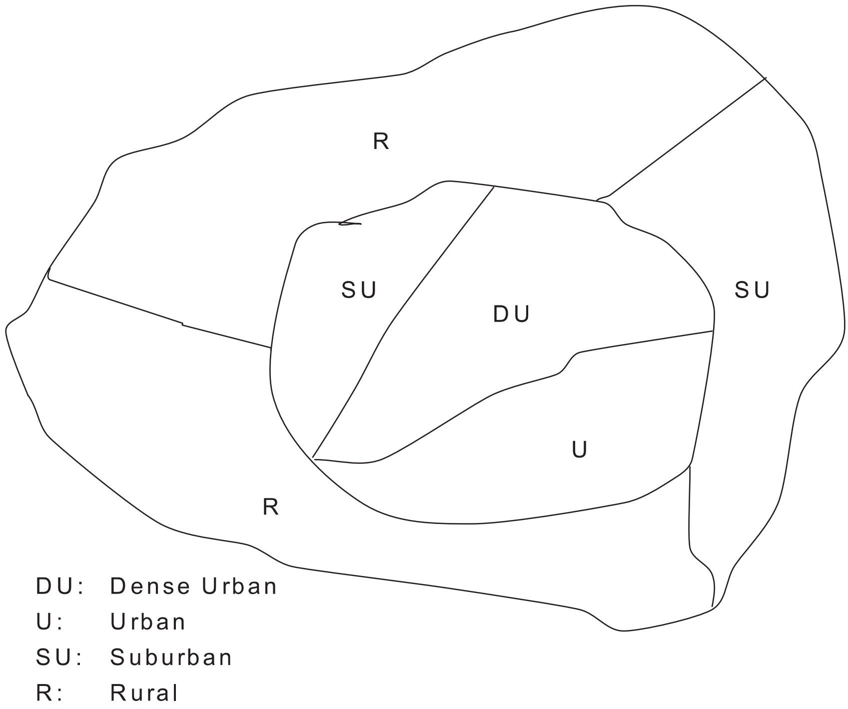


Figure 10.1 Clutter classification.

Due to the close resemblance and for the sake of calculation simplicity, the dense urban and urban categories are joined together and designated as DU/U. Hence, we now have three clutter classifications:

1. DU/U;
2. SU;
3. R.

A typical link budget for suburban morphology from Section 6.1 is presented here again. Referring to Figure 10.1, we see that there are four morphology (clutter categories) shown—dense urban (DU), urban (U), suburban (SU), and rural (R).

10.1.1 Assumptions

In this case study, as shown in Table 10.2, an uplink in-building scenario is used. As the limiting link, the budget calculation references the uplink. The maximum allowable path loss (MAPL) for the indoor uplink given in the link budget is, in this case SU, 138 dB. We will simplify by varying the margins by 6

Table 10.1
Clutter-Specific Path Losses

Clutter	MAPL
DU/U	132 dB
SU	138 dB
R	144 dB

dB among the clutter categories. To further simplify, DU and U have similar MAPL. This will effectively help us avoid a triple recalculation of MAPL. Hence, the MAPL of each clutter is as shown in Table 10.1.

The next order of business is to define marketing data, such as areas to be covered and number of households in each clutter category. Additional area and number of household assumptions are given in Tables 10.3 and 10.4, respectively, for the case study.

The rest of the technology-related assumptions in terms of parameters (such as bandwidth, frequency, and cell capacity, for example), growth rates, and other subscriber information will be given as necessary or in the coverage and capacity calculation spreadsheet that will be presented shortly.

Here is a roadmap of what is ahead:

1. Based on the MAPL, a range will be calculated for each clutter category.
2. The number of coverage sites with respect to each category is calculated.
3. The number of capacity sites with respect to each category is calculated.
4. The aggregate number of sites will be calculated, based on the number of sites calculated in 2 and 3.
5. The number of months needed to reach the break-even point, with respect to CapEx and OpEx, will be formulated.

10.1.2 Range Calculation

10.1.2.1 Range Calculation for the Suburban Clutter Category (Terrain B)

We make use of the SUI model given as:

$$L = MAPL = A + 10\gamma \log_{10}\left(\frac{d}{d_o}\right) + X_f + X_h + S$$

Table 10.2
Link Budget (Case Study)

		Outdoor (Mobile)		**Indoor (Desktop)**	
		Downlink	**Uplink**	**Downlink**	**Uplink**
Transmitter					
1	Transmitter power output (dBm)	38	30	38	30
2	Cable loss, L_c (dB)	3	0	3	0
3	Antenna gain, $G_{tx\text{-}ant}$ (dBi)	17	0	17	5
4	Number of Tx antennas, dec to dB	2	1	2	1
5	EIRP (dBm)	55	30	55	35
Receiver					
6	Antenna gain, $G_{tr\text{-}ant}$ (dB)	–1	18	6	18
7	Number of channels, Ch_{no}	16	16	16	16
8	Subchannel gain, G_{sc} (dB)	12	12	12	12
9	Noise figure, NF	8	5	8	5
10	Signal-to-noise ratio, SNR	1	2	1	2
11	Receive noise level, N (dBm/Hz)	–174	–174	–174	–174
12	Channel bandwidth, BW (dBHz)	10	10	10	10
13	Receiver sensitivity, S_{rx} (dBm)	–107	–109	–107	–109
14	Isotropic receive level, IRL (dBm) (= $S_{rx} - G_{rx\text{–}ant}$)	–106	–127	–113	–127
15	System gain, G_{sys} (dB) (= EIRP – IRL)	161	157	168	162
Link Margins					
16	Log normal fading margin, M_{ln} (dB)	8	8	8	8
17	Fast fading margin, M_{ff} (dB)	6	6	0	0
18	Interference margin, m_{int} (dB)	3	3	3	3
19	Building penetration loss, L_{bldg} (dB)	0	0	13	13
20	Total margin, M_{tot} (dB) (= $M_{ln} + M_{ff} + M_{int} + L_{bldg}$)	17	17	24	24
21	Max allowable path loss, MAPL (dB) (= $G_{sys} - M_{tot}$)	144	140	144	138

We first calculate the range, d, based on suburban clutter with MAPL, as given in the link budget above, at 138 dB.

Table 10.3
Clutter Areas

Clutter	Area (km^2)
DU/U	1,000
SU	1,500
R	2,500

Table 10.4
Clutter Households

Clutter	Number of Households
DU/U	150,000
SU	100,000
R	50,000

Given the following assumptions:
$d_o = 100\text{m}$;
S (shadowing factor) = 9 dB;
CPE height, $h_m = 3\text{m}$;
Height of BS, $h_b = 20\text{m}$;
$f = 3.5$ GHz;
Assume terrain B.

$$L = MAPL = A + 10\gamma \log_{10}\left(\frac{d}{d_o}\right) + X_f + X_h + S$$

Solving for d,

$$d = 10^{\left\{\frac{MAPL - A - X_f - X_h - S}{10\gamma}\right\}} * d_o$$

Now we need to determine A, X_f, X_h, and γ.
A is the free space path loss for the reference distance, given by:
$A = 20\log_{10}(d_{o(km)}) + 20\log_{10}(f_{MHz}) + 32.45$
$A = 20 \cdot \log 10(.1) + 20 \cdot \log 10(3.5 \cdot 10^3) + 32.45$

= 83.3

Now from the height of CPE, h_m, 3m, we can calculate the correction factor for CPE height.

$X_h = -10.8 \cdot \log 10(3/2) = -1.9$

From the frequency, *f*, we can compute X_f:

$X_f = 6 \cdot \log 10(3.5 \cdot 10^3/2{,}000) = 1.5$

Now the final variable to be calculated is γ.

$$\gamma = \left(a - bh_b + \frac{c}{h_b} \right)$$

Refer to Table 6.3 under Terrain B to substitute for a, b, and c, and h_b, given as 20m.

Then,

$$\gamma = \{4.0 - (0.007 \times 20) + 17.1\} = 4.75$$

Finally, a figure for the range, *d*:

$$d = 10^{\left\{\frac{138-83-1.5+1.9-9}{10\times 4.74}\right\}} * 0.1 = 0.95 \text{ km}$$

This range or radius of a cell seems reasonable for a suburban environment.

10.1.2.2 Range Calculation for Dense Urban/Urban Clutter Category (Terrain A)

Similarly, we now compute the range based on dense urban/urban clutter.

This time, the MAPL is 132 dB; however; the remaining parameters and assumptions remain the same, with the exception of the terrain type, which in this case is terrain A.

Given similar assumptions as above:

$d_o = 100$m;

$S = 9$ dB;

CPE height, $h_m = 3$m;

Height of BS, $h_b = 20$m;

$f = 3.5$ GHz.

Assume terrain A.

The only thing that is different this time is MAPL and γ. MAPL changes because the link budget changes due to clutter change. γ changes because the underlying model parameters (a, b, and c) depend on change.

(To review the computation and rationale of γ, refer to Section 6.3.)

Therefore, MAPL = 132

$\gamma = 5.08$

Solving for *d*,

$$d = 10^{\left\{\frac{MAPL - A - X_f - X_h - S}{10\gamma}\right\}} * d_o = 0.62 \text{ km}$$

10.1.2.3 Range Calculation for Rural Clutter Category (Terrain C)

For rural clutter, we proceed in a similar fashion, keeping all parameters and assumptions the same with the exception of

MAPL = 144 dB.

The computation for rural clutter (Terrain C) will result in

$$\gamma = \left(a - bh_b + \frac{c}{h_b}\right) = 4.5$$

Solving for *d*,

$$d = 10^{\left\{\frac{MAPL - A - X_f - X_h - S}{10\gamma}\right\}} * d_o = 1.57 \text{ km}$$

10.2 Number of Sites for Coverage and Capacity

Tables 10.5 through 10.7 present the definition of the clutters under study, while Table 10.8 presents the resulting coverage and capacity site summary.

10.3 Summary

Clearly, the design is heavily coverage driven. As can be observed from the result, the requirement for the capacity sites never exceeds the coverage sites, implying the objective area is spread out. In this case, the maximum of the two is used—coverage sites. It is possible, though unlikely in the short term, that the number of capacity sites can start to catch up and even surpass the number of coverage site beyond year 5.

Table 10.5
Dense Urban/Urban Definitions

Coverage and Capacity Planning for Dense Urban/Urban					
Coverage Area Definition	**Year 1**	**Year 2**	**Year 3**	**Year 4**	**Year 5**
Total area (km^2)	1,000	1,000	1,000	1,000	1,000
% area to be covered	15%	30%	60%	100%	100%
Area covered (km^2)	150	300	600	1,000	1,000
Penetration					
Morphology (clutter) specific number of households	150,000	154,500	159,135	163,909	168,826
Yearly growth	3%	3%	3%	3%	3%
Penetration rate	4%	6%	10%	12%	15%
Total subscribers	6,000	9,270	15,914	19,669	25,324
Data rate, Oversubscription, Revenue/Month					
Maximum data rate/subscriber (Mbps)	2	2	2	2	2
Oversubscription	50	50	50	50	50
Revenue/subscriber/month ($/month)	$39	$38	$37	$36	$35
Band Usage					
Band	3.5 GHz				
Channel bandwidth (MHz)	5				
Number of channels	2				
Bandwidth used (MHz)	10				
Coverage and Capacity Information					
Cell radius (*R*, km)	0.62				
Cell area (= 2.6·R·R, km^2)	1.00				
BTS, downlink capacity (Mbps)	25				
Number of Coverage Sites					
Accruing number of sites	151	301	601	1,001	1,001
Traffic supported (based on 25 Mbps/BTS)	3,775.0	7,525.0	15,025.0	25,025.0	25,025.0
Number of subscribers	6,000	9,270	15,914	19,669	25,324

Table 10.5 (continued)

Number of Coverage Sites					
Downlink average data rate per subscriber (kbps)	629	811	944	1,272	988
Number of Capacity Sites					
Number of subscribers	6,000	9,270	15,914	19,669	25,324
Data rate/site (Mbps)	25	25	25	25	25
Oversubscription	50	50	50	50	50
Maximum data rate/subscriber (Mbps) – given	2	2	2	2	2
Number of subscribers/site = [(data rate/site) (oversubscription)]/(data rate/subscriber)	625	625	625	625	625
Number of capacity sites = (number of subscribers)/(number of subscribers/site)	10	15	26	32	41

Table 10.6
Suburban Clutter Category Definitions

Coverage and Capacity Planning for Suburban					
Coverage Area Definition	**Year 1**	**Year 2**	**Year 3**	**Year 4**	**Year 5**
Total area (km^2)	1,500	1,500	1,500	1,500	1,500
% area to be covered	15%	30%	60%	100%	100%
Area covered (km^2)	225	450	900	1,500	1,500
Penetration					
Morphology (clutter) specific number of households	100,000	103,000	106,090	109,273	112,551
Yearly growth fault	3%	3%	3%	3%	3%
Penetration rate	4%	6%	10%	12%	15%
Total subscribers	4,000	6,180	10,609	13,113	16,883
Data Rate, Oversubscription, Revenue/Month					
Maximum data rate/subscribers (Mbps)	2	2	2	2	2
Oversubscription	50	50	50	50	50

Table 10.6 (continued)

Data Rate, Oversubscription, Revenue/Month					
Revenue/subscribers/ month ($/month)	$39	$38	$37	$36	$35
Spectrum Usage					
Spectrum	3.5 GHz				
Channel bandwidth (MHz)	5				
Number of channels	2				
Bandwidth used (MHz)	10				
Coverage and Capacity Information					
Cell radius (R, km)	0.95				
Cell area (= $2.6 \cdot R \cdot R$, km^2)	2.35				
BTS, downlink capacity (Mbps)	25				
Number of Coverage Sites					
Accruing number of sites	96	192	384	640	640
Traffic supported (based on 25 Mbps/BTS)	2,400.0	4,800.0	9,600.0	16,000.0	16,000.0
Number of subscribers	4,000	6,180	10,609	13,113	16,883
Downlink average data rate per subscriber (Kbps)	600	776	904	1,220	947
Number of Capacity Sites					
Number of subscribers	4,000	6,180	10,609	13,113	16,883
Data rate/site (Mbps)	25	25	25	25	25
Oversubscription	50	50	50	50	50
Maximum data rate/ subscriber (Mbps)—given	2	2	2	2	2
Number of subscribers/ site = [(data rate/site) (oversubscription)]/ (data rate/subscriber)	625	625	625	625	625
Number of capacity sites = (number of subs)/(Number of subscribers/site)	7	10	17	21	28

Table 10.7
Rural Clutter Category Defintions

Coverage and Capacity Planning for Rural					
Coverage Area Definition	**Year 1**	**Year 2**	**Year 3**	**Year 4**	**Year 5**
Total area (km^2)	2,500	2,500	2,500	2,500	2,500
% area to be covered	15%	30%	60%	100%	100%
Area covered (km^2)	375	750	1,500	2,500	2,500
Penetration					
Morphology (clutter) specific number of households	50,000	51,500	53,045	54,636	56,275
Yearly growth	3%	3%	3%	3%	3%
Penetration rate	4%	6%	10%	12%	15%
Total subscribers	2,000	3,090	5,305	6,556	8,441
Data Rate, Oversubscription, Revenue/Month					
Maximum data rate/ subscriber (Mbps)	2	2	2	2	2
Oversubscription	50	50	50	50	50
Revenue/subscriber/ month ($/month)	$39	$38	$37	$36	$35
Band Usage					
Frequency band	3.5 GHz				
Channel bandwidth (MHz)	5				
Number of channels	2				
Bandwidth used (MHz)	10				
Coverage and Capacity Information					
Cell radius (*R*, km)	1.57				
Cell area (= 2.6·R·R, km^2)	6.41				
BTS, downlink capacity (Mbps)	25				
Number of Coverage Sites					
Accruing number of sites	59	118	235	391	391
Traffic supported (based on 25 Mbps/BTS)	1,475.0	2,950.0	5,875.0	9,775.0	9,775.0

Table 10.7 (continued)

Number of Coverage Sites					
Number of subscribers	2,000	3,090	5,305	6,556	8,441
Downlink average data rate per subscriber (Kbps)	737	954	1,107	1,490	1,157
Number of Capacity Sites					
Number of subscribers	2,000	3,090	5,305	6,556	8,441
Data rate/site (Mbps)	25	25	25	25	25
Oversubscription	50	50	50	50	50
Maximum data rate/subscriber (Mbps)—given	2	2	2	2	2
Number of subscribers/site = [(data rate/site)·(oversubscription)]/(data rate/subscriber)	625	625	625	625	625

Table 10.8
Total Coverage and Capacity and Sites

	DU/U		**SU**		**RU**			
	Coverage	**Capacity**	**Coverage**	**Capacity**	**Coverage**	**Capacity**	**Year Coverage Total**	**Year Capacity Total**
Year 1	151	10	96	7	59	4	306	21
Year 2	301	15	192	10	118	5	611	30
Year 3	601	26	384	17	235	9	1,220	52
Year 4	1,001	32	640	21	391	11	2,032	64
Year 5	1,001	41	640	28	391	14	2,032	83

The break-even point for the case study is left as an exercise for the reader, based on the concept and formula derived in Section 6.5.

11

Preliminary Design to Benchmarking

11.1 Overview

In this chapter, various design principles and techniques of WiMAX deployment will be discussed, including preliminary to full design technique and simulation, construction of sites, optimization, and benchmarking. The main topics covered include the following:

- Rough designing;
- Reverse engineering;
- Using simulation tools;
- Designing a site;
- Selecting a site;
- Constructing a site;
- Optimizing, expanding, or improving a network;
- Redesigning a network;
- Benchmarking a network.

With each topic, important pointers critical for the design and deployment process will be discussed.

The design process starts in earnest by simply gathering design requirements and constraints. These include technical requirements and marketing input gleaned from the design requirements document and statement of work

(SOW). The design is also constrained by the technology type and hardware and software specifications.

Depending on the level of accuracy needed, the network and RF planner can use various techniques to come up with rough estimates of link budget, cell counts, and costs. Selecting from a quick-and-dirty, back-of-the-envelope computation, a spreadsheet-driven, detailed computation, or a full-blown prediction tool and measurement-data supported design, the planner can choose the levels of accuracy and expeditiousness, making tradeoffs as necessary.

Assuming the results are needed right away, and the accuracy is not that important at the initial stage, a quick estimate can be made by means of a back-of-the-envelope computation approach. (For detailed design guidelines, please refer to Chapters 6 and 10.)

11.2 Back-of-the-Envelope Link Budget and Area Estimate

A very quick link budget can look like the one in Table 11.1.

According to the rough estimate in Table 11.1, it is clear to see that the limiting link is the uplink, on account of limited SS power. For this reason, all calculations reference this link. Assuming a balanced link, we can see that by varying the transmit power, the cell counts vary dramatically. Now from our previous study, we know that

$$L = MAPL = A + 10\gamma \log_{10}\left(\frac{d}{d_o}\right) + X_f + X_h + S$$

where A is:

Table 11.1
Bare-Bones Link Budget

Downlink		**Uplink**	
BS Tx power	50 dBm	Mobile Tx power	30
Antenna gains (BS + SS)	17 dBi	Antennae gains (SS + BS)	17 dBi
Cable loss	–3 dB	Cable loss	–3 dB
EIRP	64 dBm	EIRP	44 dBm
aspnumFade margin	–15 dB	Fade margin	–15 dB
Net power	49 dBm	Net power	29 dBm
SS Rx sensitivity	–100 dBm	BS Rx sensitivity	–109 dBm
MAPL	149 dBm	MAPL	138 dB

$$A = 20 \times \log 10(0.1) + 20 \log 10\left(3.5 \times 10^3\right) + 32.45$$

S about –10 dB

A represents the free space path loss at a reference distance, 0.1 km, including a loss due to frequency at 3.5 GHz and a loss constant of 32.45 dB (using units of km and MHz, respectively.) A rough estimate of *A* is about 83 dB.

Then it can be seen that A + S = 83 + 9 = 92 dB (X_f and X_h are negligible).

Then MAPL = 138 (from our abbreviated link budget in Table 11.1), loss slope ($10 \times \gamma$) assumed is 38 dB, γ, assumed 3.8 driven by the reflective, obstructive, and/or diffractive nature of the environment.

$$\begin{aligned} &138 = 92 + 38 \times \log_{10}\left(d/d_o\right) \\ &d = 1.6 \text{ km} \end{aligned}$$

Let's say that we vary the transmit power by +3 dB.

Then we can determine the radius and area of the cell site and ultimately the cell counts across the network.

$$+3 \text{ dB} = 38 \log_{10}\left(\frac{R_{new}}{R_{old}}\right)$$

where R_{new} and R_{old} are the new and old radii, respectively.

$$R_{new} = (3/38) \times 1.6 = 1.9 \ (R_{old} = 1.6 \text{ km})$$

Increasing the transmit power results in 18% more than the old radius.

Calculating the differences made in the areas, taking into account the old and new radii:

$$\begin{gathered} AREA_{new} / AREA_{old} = \pi \times R_{new}^2 / \pi \times R_{old}^2 = (0.1)^2 / (1.0)^2 = 1.4 \\ AREA_{new} = 1.4 \times AREA_{old} = 1.4 \times \pi \times (1.6)^2 = 11.26 \text{ km}^2 \\ AREA_{old} = \pi \times (1.6)^2 = 8 \text{ km}^2 \end{gathered}$$

Assuming the coverage area for this type of cell is roughly 100 km^2, using the new radius or area, we need 100/11.26 = 9 cell sites. In contrast, if we were still using the old radius or area, we would need 100/8 = 12 cell sites.

Hence, increasing the transmit power by 3 dB (or alternatively decreasing a loss, which can be a cable loss) results in the reduction of 9 cell sites in an area of 100 km^2. This translates to

$$(9-12)/12=-25\% \text{ (a 25\% reduction in the number of cell sites)}$$

11.3 Reverse Engineering

Supposing, earlier, that a portion of the network were already deployed, and there were a need to characterize that network, either to build a network similar to it, verify if certain quality performance indicators adhere to requirements, modify certain aspects of the design, or for a variety of other reasons there is a need to deduce the performance of the system. Then a simple reverse-engineering process can be followed to deduce, say, the link budget, average cell radius, quality, or coverage confidence. Such actions will be necessary, owing largely to the unavailability of design documentation or measurement records.

The centerpiece of this effort is acquiring measurement data. If other types of data, such as network management statistics (NMS) or switch statistics (dropped calls, access failures, bit error rate (BER), and so on) are available, then they can be used to corroborate the measurement data in case of uncertainty or to supplement it where data could not be collected. To begin the process, sample sites representing the various morphologies are selected as a candidates for measurement testing. Ideally, at least two or three sites per morphology is selected. (That is, two to three from dense urban, two to three from urban, and so on.) Drive test routes are prepared for continuous wave signal reception. Each site belonging to the same morphology is processed separately. If need be, sites from the same morphology can be statistically averaged to remove any biases. Depending on the type of post data processing and spreadsheet software used, a number of critical pieces of information can be extracted. Refer to Figure 11.1.

For example, from this analysis the following information can extracted:

- Approximate cell radius for a given morphology;
- Path loss at the cell edge;
- Reference distance (x_1) below which a free space path loss applies;
- Cell edge (at x_2) corresponding to receiver sensitivity;
- Cumulative probability function for given morphology data points;
- Coverage probability at the cell edge and over the entire area can be determined, from which fade margin can be deduced;
- Transmit power.

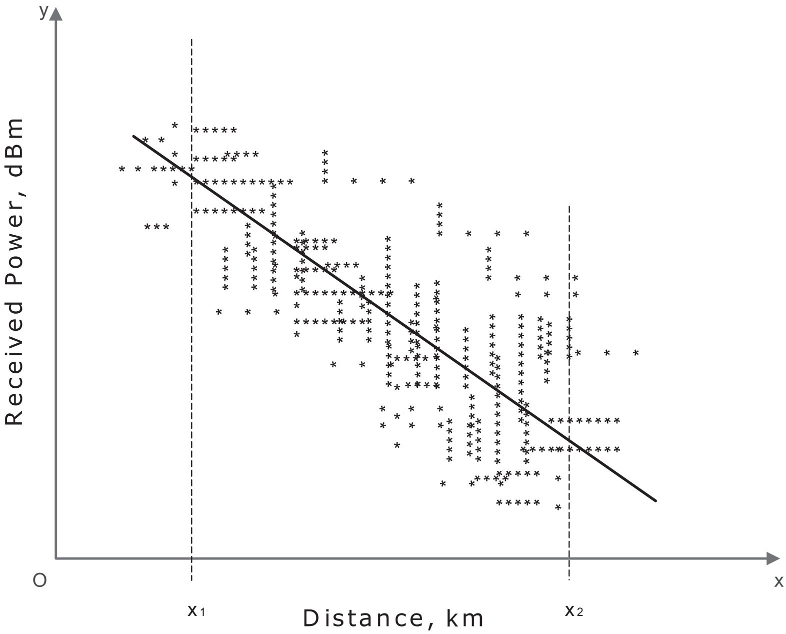


Figure 11.1 Reverse-engineering link and performance.

So, in short, the entire link budget can be reconstructed, including the loss slope for the given morphology. This analysis bears a lot of similarity to the process used for model tuning, which will be discussed in Section 11.4.

11.4 Model Tuning

Model tuning, an important part of the design process, involves a propagation model adjustment. While propagation models do a fairly good job of computing power losses from a transmitter to any point in study region, they have to be told (in terms of a correction factor) which clutter or morphology type is being studied. Propagation models do take the underlying terrain information to factor into the computation. However, unless correction factors with respect to each morphology are given by way of constant numbers or via another layer, similar to terrain information or land-use clutter information, the predictions will be wrong. Clutter information or morphology type is an indicator of the type of man-made objects that are present in the environment. For example, dense urban would have crowded and tall buildings, as well as many moving cars, trucks, and other stationary and moving objects that are detrimental to RF propagation. Most propagation algorithms set aside a constant value just for this correction factor. Then, when editing the sector information, not only is there a provision for electing the type of propagation algorithm, but also for specifying the clutter loss represented in dB. (It is true that representing the entire sector with one loss value can be erroneous without considering the look angle of the antenna. Certain industry tools have a way around this problem, by specifying the intervening obstructions.)

Briefly, the model tuning process entails selecting two to three candidate sites per morphology, preparing well-considered drive test routes, and setting up a transmitter site with power level, antenna type, and height similar or as close as possible to the one that is going to be in the actual site. However, if there are variations for any reasons, the measured data must be adjusted to match the actual design parameters. Keep in mind at this stage of the design that no site is up and transmitting, hence the need to set up a transmitter.

Collecting the measurement data mainly consists of received signal strength (given in dBm) per location, as given in latitude and longitude information. For collecting data, a setup is used that consists of a mobile phone, GPS receiver, and laptop equipped with software capable of collecting the received data and GPS information, as well as storing and or displaying it on the monitor.

The outcome of this exercise will also yield the all-important cell radius information corresponding to each morphology or clutter. This is critical information that tells the designer, based on reality, how far to space the cell site, at least for the coverage condition. If the design is done well, the radii of the cells must come close to the link budget developed. Refer to Figure 11.2, which shows the processed data and path loss versus distance.

It is shown that after the data points are fitted with a mathematical relationship that relates distance to path loss: L (path loss) = a(intercept) + b(loss slope)$\log_{10}x$. As can be seen, at the reference distance $x_{1,}$ L reverts to a. This occurs at $x_1 = 1$, which can be scaled accordingly. It is important to note that the loss below this reference point follows a free space path loss rule. If one compares this simplified model with the SUI model, it can be see that the great majority of a, as given in that model, consists of free space path loss. However, here we are sort of force-fitting the rest of the terms in the SUI model (namely, frequency and SS height correction, as well as lognormal fade) into a.

On the other hand, if we compare L (path loss) = a(intercept) + b(loss slope)$\log_{10}x$ with another model, namely, the power law model, they are very similar.

L(loss) = $10n.\log_{10}(d/d_{ref}) + L_{ref}$, where n is the path loss exponent, and L_{ref} is a loss based on free space at a reference distance. The only small difference is that the $\log_{10}(d/d_{ref})$ only computes loss at distances over and above the reference distance, d_{ref}. Hence, to prevent our data-point-fitting formula from counting the reference loss twice, we must compensate for it in the processing.

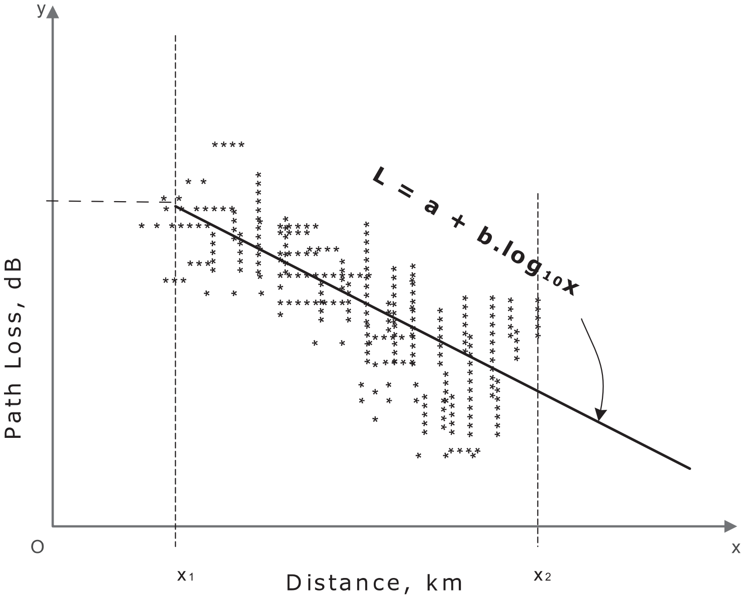


Figure 11.2 Drive test for model tuning.

b (equivalent to 10*n* in the power law model) is a very important variable that is a signature value for the morphology under consideration. Dense urban environments have a greater value than urban, suburban, or rural regions.

11.5 Model Tuning Using a Prediction Tool

In some instances, where the prediction tools are not equipped with model-tuning features or in cases where there is incompatibility with measurement data and the type of data that can be imported into the prediction tool, the foregoing discussions apply very well. However, in many situations the prediction tool is equipped with modules that allow the importation of measurement data, so that prediction and measurements can be compared and the differences can be input as correction factors. Each tool works slightly differently, and the respective manuals must be consulted. The following is a general concept that can apply for most tools:

1. The number of data points must be at least 30 to do any meaningful statistical analysis.
2. Cut out points closer than 100m to the transmitter or further than that where the signal is lower than the receiver sensitivity or wherever there are sudden spikes.

3. The mean of differences of measured and predicted means should be close to zero.
4. The standard deviations of differences of measured and predicted standard deviations must be as small as possible—less than 8 dB.
5. Data points should fit the following form: L (path loss) = a(intercept) + b(loss slope)$\log_{10}x$.

11.6 Site Design Using a Prediction Tool

Equipped with design requirements, marketing input, design objective, hardware specification, link budget, cell radii, approximate cell counts, clutter-specific losses and a tuned model, one is almost ready to start radio planning using a prediction tool.

Some use the planning tool to plop sites over the coverage area almost indiscriminately, with an eye toward just satisfying the coverage need from a theoretical vantage point. However, this approach, for the most part, is ill-advised and should be minimized. First, the theoretical placing of sites just anywhere can result in the wrong location and even significant variation from that location will likely change the RF coverage significantly. And more important, that location may not be realized in the site-acquisition process. Hence, this approach generally must be discouraged.

A more productive and efficient approach is one that involves researched and real sites to serve as anchor sites, around which other less semi-theoretical sites with multiple alternatives are designed into the network to fill in the remaining gaps. While the anchor sites are based on real structure locations and a higher likelihood of securing a lease, the fill-in sites can change, and a few alternatives must be prepared ahead of time. The anchor sites are based on buildings, water tanks, or collocation of existing antenna towers, all of which have improved odds of being available for leasing. So, in short, the radio planner must go out and gather latitude and longitude information of the possible and real site locations, based on the coverage objective of the design. This can include:

- Latitude and longitude;
- Address, cross street;
- Name of business;
- Type of structure;
- Approximate height;
- If possible, phone numbers of owner or representative.

A database of anchor sites should be created and ranked according to RF suitability and the probability of securing a lease. In order to rank them from an RF perspective, several RF study iterations should be done. The first iteration should be one that includes all the anchor sites. Subsequent iterations will successively turn off sites in the coverage studies that appear to be redundant. Then a variation of these studies will be created, turning on the sites turned off in the previous studies and turning off those that were turned on.

Anchoring design on real sites has one great obvious benefit: time saved that would have been lost during redesign when invariably the theoretical design fails to align with the coverage objective. The final goal is building a working site location that meets the coverage objective. In other words, considerable time would be lost in adjusting theoretical sites to fit into the actual design and make it ready to be built. On the other hand, it is also true that some finite time will be required to find real sites to which the design can be anchored. However, the time lost for design and redesign to conform all theoretical sites to final design would be considerably higher. In any event, coordinating the computer design with an actual visit to the field or to site-acquisition personnel expedites the design and deployment process.

Ideally, you want the best location from an RF perspective that you can lease with minimal to no opposition from the neighborhood when the time comes to actually construct it. Well, how do you know if a site location is ideal from an RF vantage point? Mainly you want a clear view overlooking the coverage-objective locations. If, for example, a rooftop of a building is the prospective location, go up to the rooftop where the antennas are supposed to be located, and verify if your coverage objective is in clear view. For example, if it is a highway intersection you wish to cover from that location, then verify that the intersection is in clear view from the antenna location. Imagine the antenna propagating vertical and horizontal beamwidths, and make sure no nearby object or a building's edge or parapet obstructs the propagation. If necessary, you should take back a sketch and measurement of the area, along with the antenna height, to your office and determine the proper clearance and angles.

If the prospective location is a tower being used by other service providers, you can rest assured it is a good location. However, you should determine where your antennas are going, whether higher or lower (or in the middle) than the other providers. Of course, frequency and distance (both vertical and horizontal) separation must be verified. (The guideline for vertical and horizontal separation of antennas is covered in Section 5.5.)

The height of the site location is critical. While a very tall site is nice to meet the coverage objective, it could be a source of a serious interference. If it so happens that it is the only location you can have in that area, you can try tilting the antenna down quite a bit, but even that may not work. Here is a brief guideline:

- Average building height (cheap for licensing and available for zoning, but limits the antenna range);
- Above-average building height (expensive, best transmissions and reception, and closest to LOS);
- Below building height (cheapest, increases network capacity because the cells from short-height antennas are small).

When you select a rooftop, where you actually locate the radio is critical. If the distance between the antenna location and the radio is long, then the losses incurred can be prohibitive. In some cases, to counteract the losses, you might elect to use a bigger gauge cable. In any case, the link budget must be revisited and verified. The computer simulation must also confirm that the coverage objective is met.

If your coverage objective is a bending highway, consider locating the tower (antenna) right around the bend and try installing two sectors, each sector overlooking a side of the bend.

Going back to the design process, it is a known fact that not all the sites in the design iteration can be real sites that will eventually make it to the final configuration. Some, despite their suitability from an RF perspective, cannot be leased or there can be other restrictions in the area, and the engineer must resort to the second-best or even third-best option. Design fallbacks must be prepared to work around such problems. The fill-in (or theoretical) sites, those that are not part of the real or anchor sites, must allow for a variation in location. In other words, since the absolute location of the theoretical site design is virtually impossible to secure, the design must cover a variation of the location to within 25% of the coverage radius. Site acquisition must try to find a site within 25% of the circle, and the design must support that. The design must also allow for a strategy to avert the worst-case scenario in which multiple adjacent sites can only be realized away from each other, thereby creating a coverage hole. This concept is best illustrated by Figure 11.3.

The point of Figure 11.3 is that the worst-case scenario is where adjacent search rings are either outside or at the edges on the opposite side, so as to open an RF hole. In any event, the radio planner and site acquisition department should work hand-in-hand to expedite the design and leasing process.

The Site Qualification Form in Figure 11.4 facilitates the site search and acquisition process.

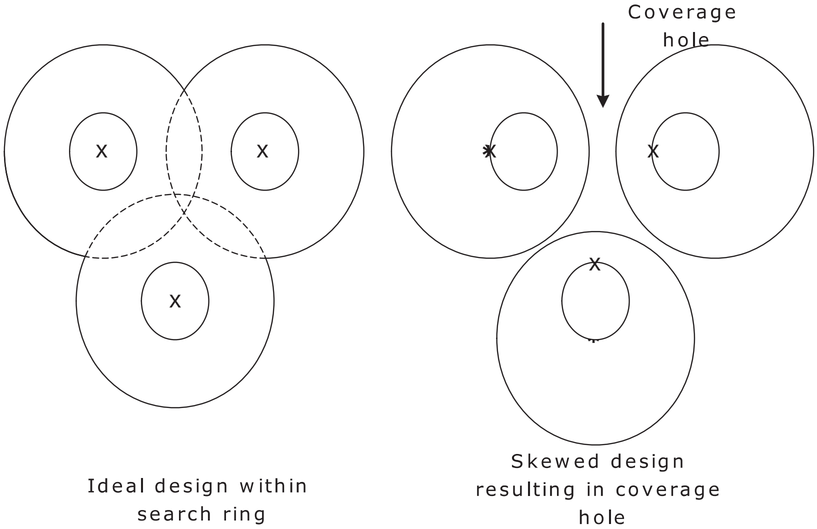


Figure 11.3 Site acquisition based on ideal and worst-case locations with respect to the search ring.

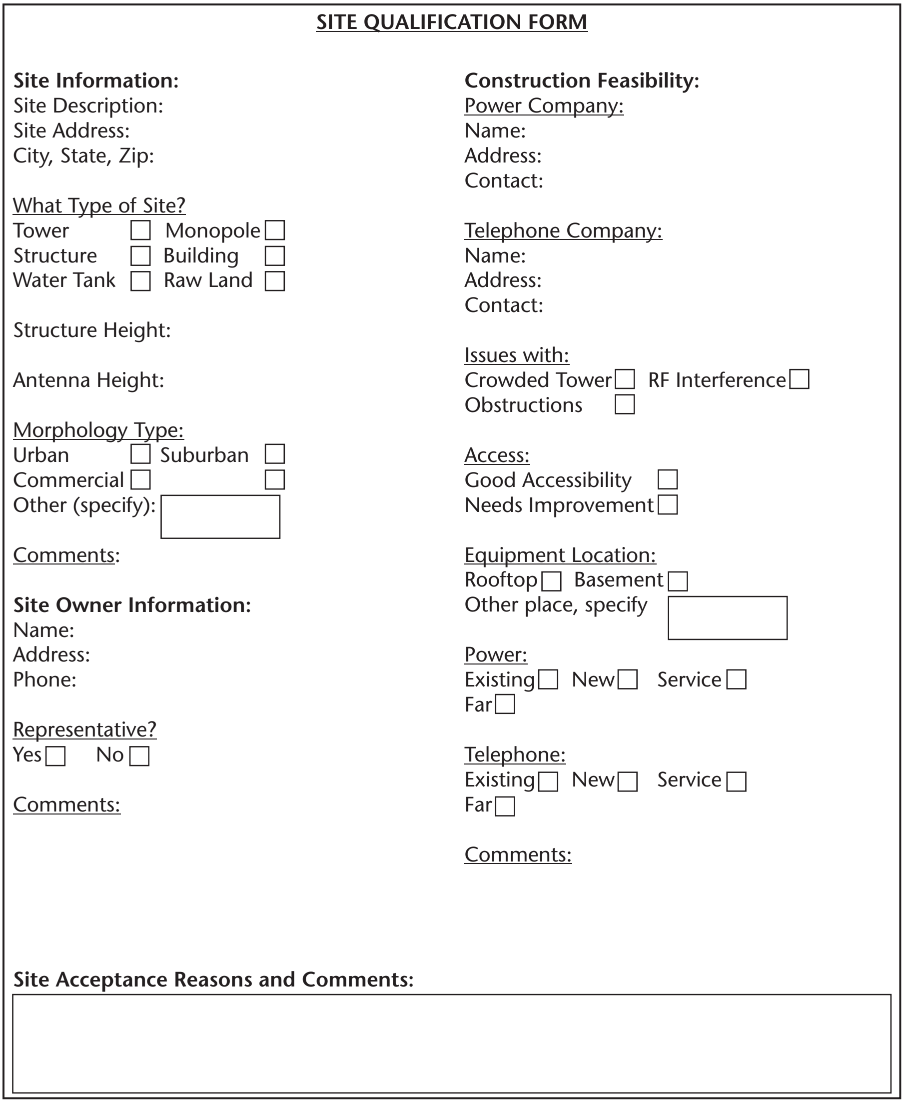
11.7 Site Qualification

Prior to site selection, the form in Figure 11.4 can be used to rank, evaluate, and qualify a site from RF, leasability, constructability, and feasibility standpoints. The form organizes the essentials necessary for a successful network deployment.

Please note that it cannot be overemphasized that, prior to a major investment and site development, the deployment frequency band must be tested for interference pollution thoroughly. Pronounced, persistent, and perennial interferences, unless they can be cleared from the band in time, may not be worth the investment in the site location. This and other sources of interference testing concepts and methods are presented in Chapter 5.

11.8 The Use of Planning Tools

Industry standard planning, prediction, or simulation tools work in similar way. The algorithm engine typically computes and assigns a path loss (or received power) from a transmission point to any point in the deployment region, as defined by the pixel size and dictated by terrain and land-use data. In order to make the study region more familiar to the user, highway and street map layers, color coding, and differentiation of various power levels (or other study parameters) are superimposed and displayed on the monitor. The transmitter antenna



SITE QUALIFICATION FORM

Site Information:
Site Description:
Site Address:
City, State, Zip:

What Type of Site?
Tower ☐ Monopole ☐
Structure ☐ Building ☐
Water Tank ☐ Raw Land ☐

Structure Height:

Antenna Height:

Morphology Type:
Urban ☐ Suburban ☐
Commercial ☐ ☐
Other (specify): ☐

Comments:

Site Owner Information:
Name:
Address:
Phone:

Representative?
Yes ☐ No ☐

Comments:

Construction Feasibility:
Power Company:
Name:
Address:
Contact:

Telephone Company:
Name:
Address:
Contact:

Issues with:
Crowded Tower ☐ RF Interference ☐
Obstructions ☐

Access:
Good Accessibility ☐
Needs Improvement ☐

Equipment Location:
Rooftop ☐ Basement ☐
Other place, specify ☐

Power:
Existing ☐ New ☐ Service ☐
Far ☐

Telephone:
Existing ☐ New ☐ Service ☐
Far ☐

Comments:

Site Acceptance Reasons and Comments:

Figure 11.4 Site qualification form.

locations, legends, latitude and longitude are optionally displayed on the monitor.

To perform propagation study on a typical simulation tool one needs the following preparation:

1. Terrain data (typically 30-m resolution);

2. Land-use data, representing a loss for each morphology classification. Alternatively, such losses can be input directly on the propagation model on a per-sector basis;
3. Definition of study area, specifying the four corners using latitude and longitude;
4. Highway and street map layer to overlay the study area;
5. Manufacturer's specifications for BS and SS, such as transmit power and receiver sensitivity;
6. Antenna information for both the BS and SS;
7. Link budget information (fade margin, interference, noise figure, other margins, and so on);
8. Specification of propagation model.

A typical and simple study involves first placing sites in the defined study area, editing the radio transceiver, and adding sectors, antennas, and losses. The next step is running the study desired—coverage, interference, and so on. Next is, based on the outcome, modifying antenna types, tilt angles, or heights, changing transmitter powers or adjusting losses, and running the study again until objectives are met. Figure 11.5 summarizes the study procedure.

11.9 Site Construction

Once the coverage runs for the network (even for parts of the network) result in an optimum outcome, the acquisition process begins in earnest. Refer to the Site Qualification Form to organize the site acquisition process and understand the critical items required. Based on the information gathered and a review by the design and site acquisition team, a decision is made to lease sites. The review includes such critical issues as RF suitability and site constructability.

The site owner and the WiMAX service operator sign a lease. The service operator provides a drawing of cable runs, radio/antenna weights, power terminals, and other important information. Right after the site lease is finalized, representatives from various specialties, such as electricians, riggers, integrators, site owners, and, of course, a representative of the service provider will visit the site to assess what it will take to ready the site location for system installation.

After the site location is readied for construction, a licensing application will be filed with entities such as the FCC. The rest of the work—electrical, infrastructure, cable, antenna installation radio—should be coordinated as well. Also, certain municipal codes may require permits for radios or antennas to be

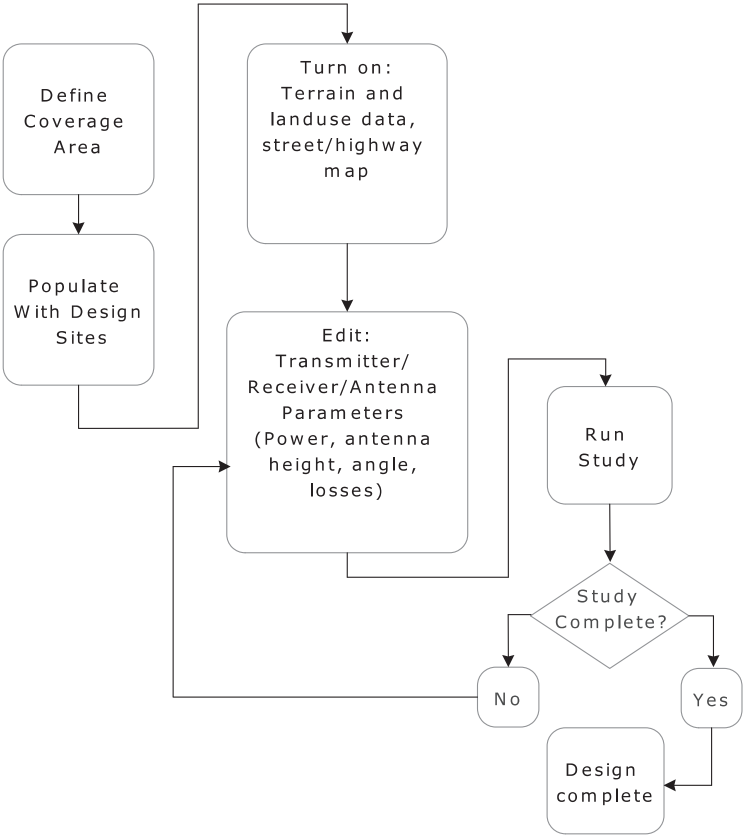


Figure 11.5 Simple steps to using an RF planning and prediction tool.

installed within their boundaries. The operator must be also ready, if necessary, to appear before a zoning committee to defend the site installation and seek approval.

11.10 Optimization

Once a number of sites, or a cluster of sites, have been constructed and brought online (even before each site is turned on), the default parameter setting follows. These parameters include transmit powers and/or gains, a neighbor list, and handoff and other parameters. From the air-interface standpoint, the two most critical performance indicators are how soon a user can attach on the system and, once a user is on the system, whether he or she can stay on it as long as required. These indicators are known as:

- Access failure;
- Dropped calls,

> As far as the network is concerned, the critical performance indicators are how quickly one can get connected to the network servers and how efficiently one is served while still on the network (delay, throughput, and jitters): the time necessary to be served or for the Web site to fill up is delay; a file transfer rate is throughput; variation in packet-arrival time, especially in VOIP, is jitter.
>
> For WiMAX-specific optimization, there are optimization tools available in the market for real-time data collecting, troubleshooting, maintaining, optimizing, and benchmarking mobile WiMAX. Such tools can log and decode PHY and MAC layer information and packet data. These tools also perform multichannel continuous wave (CW) received signal strength indicator (RSSI) measurement, preamble index detection, preamble RSSI, and carrier-to-interference and noise ratio (CINR). For example, a product called YellowFin WiMAX Analyzer (www.bvsystems.com) allows complete spectrum analysis in the range of 2 to 5.9 GHz, packet demodulation, RSSI measurement, multipath and CINR analyses, and interference source pinpointing.

which are checked to see if they are within acceptable limits. The tests can be carried out through an automated call-generation recording mechanism with respect to these two critical parameters, as well as others that require such simulations. The simulation should be done at various load conditions. The load can be on either hardware, software, both, or even actual users. Generally, the logging of the data is done at the switch, especially for those parameters requiring reverse-link data logging.

Coverage, capacity, and interference must also be checked to verify they meet the design objective. It should be noted that the presence of interference invariably reduces capacity. The general and favorable approach to mitigate interference is to either turn off or diminish the strength of the interference first and then, or in addition, create a dominant server signal in the target area where interference was not wanted. Generally, turning off or diminishing interference levels entails either downtilting the antenna/antennas in the case of multiple interference sources, or at the switch, reducing the gains belonging to the offending source, or both. Interference issues, along with the attendant capacity issues, can be solved using this simple technique. However, in most other instances, the addition of BSs solves coverage, capacity, and interference issues.

Frequency-related interferences are usually identified from measurement data, and they can usually be solved by retuning the frequency plan. The nice thing about WiMAX technology is the inherent fractional frequency reuse capability. Especially around the fringes of a cell where interferences [or low carrier to interference (C/I)] from other cells are at their highest, a fraction of the total available subchannels is allocated in such a way that adjacent cells will operate on different sets of subchannels. At the same time, to maximize the frequency reuse factor, users closer to the BS use all the subchannels or the available channels. This method requires systemwide planning and coordination and the scheduling of subchannel allocation tables. This is tantamount to dynamically changing the frequency reuse factor during a frame. C/I issues typically arise from insufficient reuse distance separation of the same frequency, which can be measured and verified. WiMAX can also be susceptible to interference from outside systems using similar unlicensed spectra, for example.

11.11 Expansion and Quality Improvement of a Network

Expansion simply involves the addition of more sites into a newer area, although the act of meeting capacity demands can be deemed expansion as well.

Expansion into a new area is basically identical to designing a brand new site, and all the design steps apply. Quality improvement means enhancing the performance of the network. Generally performance indicators of the traditional wireless network are access failure and dropped calls. When it comes to WiMAX, a few others can be added, such as delay in filling up a Web site, number of packet losses, delay in VOIP, and jitters.

11.12 Benchmarking

Benchmarking is a means of taking a snapshot of the network in order to create a baseline for future improvements. By identifying and documenting the performances and deficiencies of the network [in terms of key performance indicators (KPI)], the network is set up to quantify any future improvements, taking into account changes, resolutions, betterments, or upgrades. The larger picture, over the years, will show how the network is evolving toward fulfilling the quality and capacity objectives. The benchmarking process can come any time after optimization when the system is stable.

The three most major sources of data for performing analyses are:

1. Network management system (NMS), or switch statistic;
2. Drive test or measurement;
3. Customer or personnel input.

A hardware and software system, NMS is a means of connecting to the switch so that data pertaining to the performance of the system can be extracted, displayed, analyzed, logged, and printed.

Using a laptop computer with applicable software and a GPS receiver, drive-test-based measurement ties received signal information to location information and similarly displays and or logs data as needed.

Before collecting data for benchmarking, the following pointers should be observed:

The weather condition, time of day, day of the week (weekday or weekend), and season should be recorded, It is strongly recommended that subsequent tests be done in as similar conditions as possible. Foliage conditions, for instance, make a huge difference in performance, sometimes up to 10 dB, depending on the types of trees. New structures and buildings that went up in the area must be noted carefully. Benchmarking and network performance gauging should also take into account the type of SS used, as that affects the link-budget performance due to transmitted power and receiver sensitivity issues. Typically, it is recommended that the benchmarking and network optimization process be done with a calibrated and standard phone. A similar phone has to be used for successive tests, it goes without saying.

The objective of benchmarking, as underscored above, is that the first analysis of each system sets a reference point (benchmark) for that particular network. Future analyses take place at intervals such that the system can mature and implemented changes are reflected. The different areas evaluated in benchmarking are based on the following major parameters and KPI:

1. Traffic performance:
    a. Dropped call rate;
    b. Handoff failure rate;
    c. Ineffective attempt rate;
    d. Congestion;
    e. Busy hour;
    f. Daily average performance;
2. Reliability and quality:
    a. Coverage (signal >= receiver sensitivity);
    b. Capacity (radio, link, and core network);
    c. Cell area or edge reliability (service availability, dependent on link budget and fade margin).
3. Customer satisfaction:
    a.Reported;
    b. Solicited or simulated;
    c. Volunteered.
4. Quality of service (reliability of communication link, generally affected by slow fading):
    a. Mean opinion score (MOS)—done in a lab setting;

   b. Bit-error rate (BER);
   c. Frame-error rate (FER) (from drive test).
5. System- or network-related performance:
   a. Delay (packet taking too long, Web or VoIP related);
   b. Jitters (time variation between packet arrivals, mostly for VoIP);
   c. Throughput (the average rate of successful message delivery, for example, download).

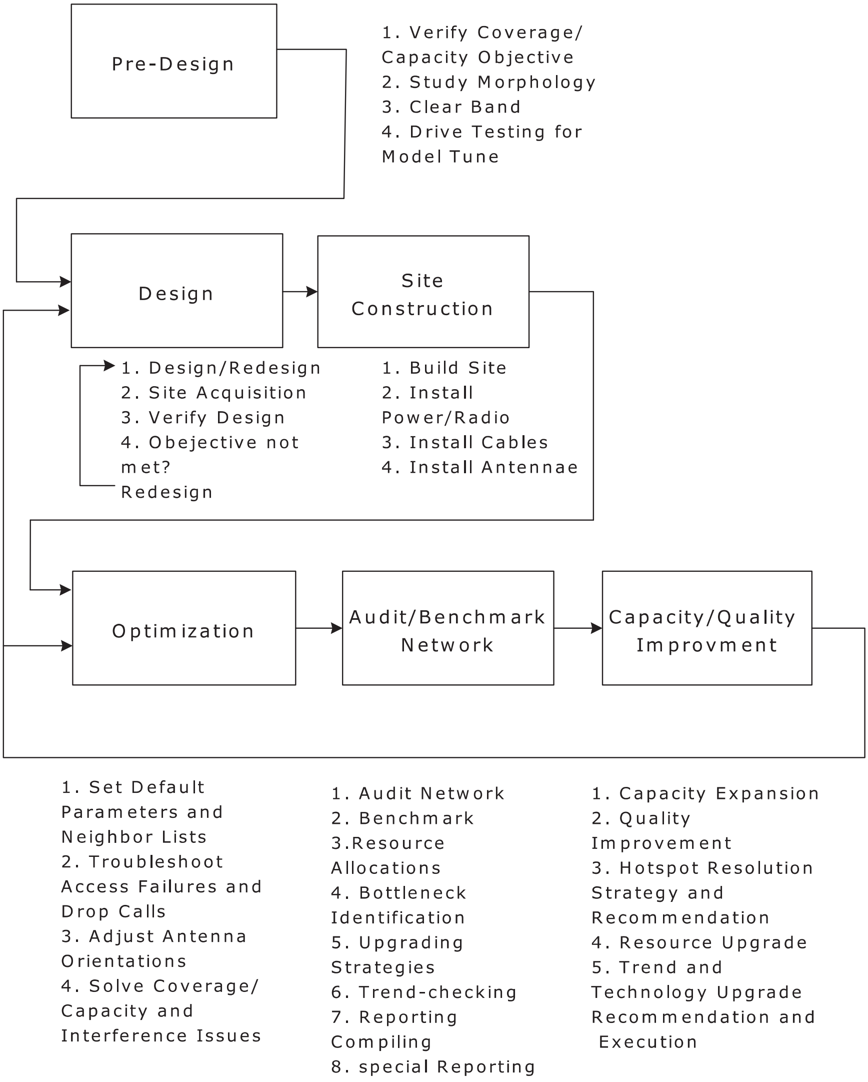


Figure 11.6 Radio network deployment cycle.

The data pieces are collected, compiled, and analyzed per sector, cell, cluster, hot spot, and important landmark where revenues are. Spreadsheets, histograms, graphs, plots, and maps are produced to show the state of the network at a glance. A comprehensive document, with descriptions, spreadsheets, graphs, and plots with indexes is produced. More importantly, the accompanying report should include a detailed explanation of the discrepancies, as well as a recommendation to ameliorate the network or solve the problems uncovered. An executive summary of the finding is also written, and the management is apprised of the status of the network.

For each network performance test, this is repeated. The only difference is that the successive tests should also reflect the performance differences and, in some cases, its decline.

The foregoing discussions, starting from predesign to deployment, to expansion, to benchmarking are summarized in Figure 11.6.

Part IV
Network and Security

12

Fundamentals of WiMAX Network

12.1 Introduction

WiMAX is both a physical (PHY) and media access control (MAC) layers specification. Most of our discussions in the previous chapters covered the PHY layer aspects—the air interface and channel characteristics. One of the stronger features of WiMAX is quality of service (QoS), and it is handled, for the most part, at the MAC layer level. For the sake of completeness and its importance, the fundamental aspects and functions of the MAC layer will be discussed in this chapter.

The significance of higher layers from the standpoint of radio resources management, mobility, handover, roaming, and overall network function, as related to Internet Protocol (IP), makes the network aspect of WiMAX important.

Secured communication is another important reason for discussing the network aspect of WiMAX. WiMAX is important, not only as a consequential technology for its various strong features, but also because it provides a more secured communication, compared to similar technologies, such as WiFi.

It should be pointed out that this discussion will be a fundamental approach to the architecture definition of WiMAX, preceded by background information on IP-based networks and protocols.

As indicated from the outset, our discussion of the various applications of WiMAX includes Web browsing, voice over IP (VoIP) (an alternative to cable and DSL for last-mile wireless broadband access), and point-to-point (fixed LOS) and point-to-multipoint (fixed LOS/NLOS) full mobile citywide applications (also called wireless metropolitan area network (WMAN)).

It is important to underline that WiMAX is largely an IP-based application working on the network layer, except when bridged, as in ATM and Ethernet protocols.

As a prelude, in this chapter, open system interconnection (OSI) and Internet protocol (IP) are discussed as a reference to and for comparison with WiMAX network layers. The WiMAX frame structure, as it relates to the data link (DL) layer, as well as how downlink and uplink frames are organized, are explained next. Finally, QoS and the fundamentals of VOIP are discussed.

We begin this chapter by defining the OSI model as a basis of the IP model, followed by more discussion of individual layers and leading up to the WiMAX model and network architecture and how it fits and compares with the IP network or end-to-end network of the Internet.

The PHY layer, the air interface, characterizing the medium of communication, has been dealt with rather adequately in Chapters 2 through 10, and now, with a few exceptions, we delve into the higher layers, starting with the DL.

12.2 OSI Models and IP-Based Networks

The OSI model is an important model as the basis for IP-related computer communication, from which the WiMAX model is derived. WiMAX, as an IP-based technology, leverages itself from tremendous advances made in computer communications and on the Internet.

Before two computers can communicate, there has to be an accepted common language and handshaking protocol, there has to be a medium through which the communication takes place. The hardware interface and software types and versions must be established. The communication must be understood by both parties; what addresses to use, whether and how to correct any errors, and how to handle collisions if and when they occur. (And, in case of WiMAX, via the MAC layer, negotiate QoS.) The OSI model protocol not only provides the rules, but also the structure around which IP-based communication is established. Refer to Figure 12.1. The IP model is nothing but an OSI model without layer 5 (session) and layer 6 (presentation), which are hardly ever used. Alternatively, layers 5 and 6 can be viewed as layers merged into layer 7 (application layer). Hence, the IP model (sometimes called the TCP/IP model) has only 5 layers. Refer to Figure 12.2, showing the IP protocol stack [1].

Each layer of these models can be regarded as a protocol or set of rules governing the function of that particular layer. For example, the session layer, layer 5 in the OSI Model, is a protocol defining the setting and tearing down of sessions connected across a network. This protocol, for example, defines a set of rules governing a session of a video download. One of these rules can then define

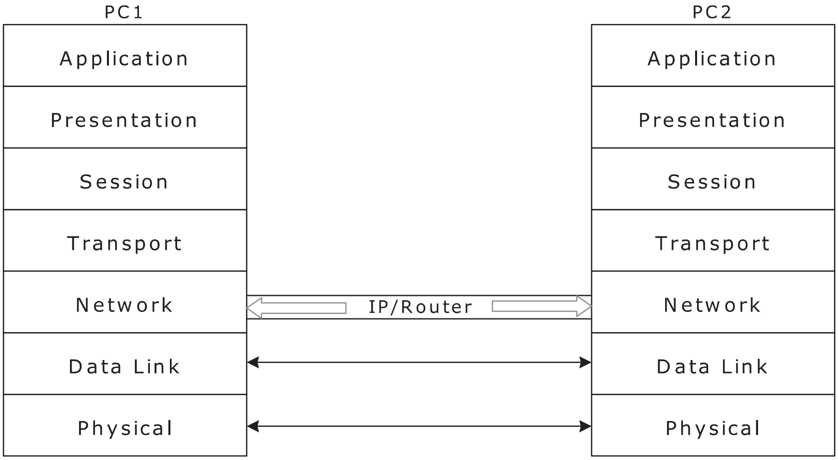


Figure 12.1 OSI model.

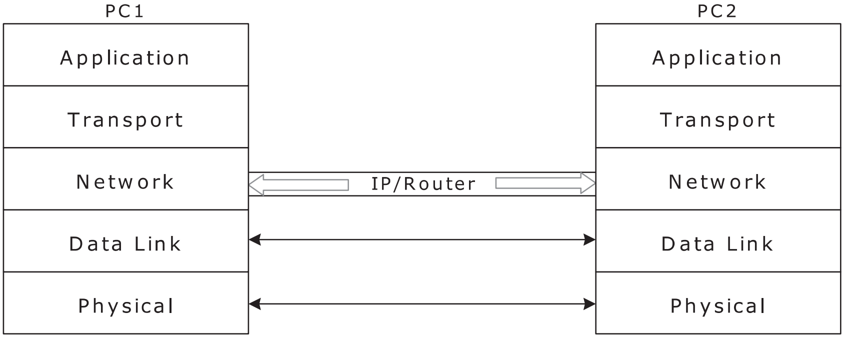


Figure 12.2 IP model.

the procedure of how to resume a download session at the point left off, or after power failure, for example.

Similarly, the presentation layer, layer 6 (also called the *syntax layer*), converts network formats to a format acceptable to the Application layer and encrypts the data to be sent over the network. Well-known formats for text, audio, images, and video are:

- ASCII—text;
- Quick Time—audio;
- Joint Photographic Experts Group (JPEG)—images;
- Quick Time and Motion Picture Experts Group (MPEG) —video.

12.2.1 PHY Layer

The PHY layer defines the type and characteristics of the medium, such as cable, air interface, bit rate, voltages, or physical interface. The data ready to be transmitted through the medium, the PHY layer, is an encapsulation of higher layer protocols: PHY = {DATA BITS + PHY layer HEADER + (ALL HIGHER LAYER PROTOCOLS (i.e., Application + TP + NW + DL)}.

12.2.2 Data Link Layer

The data link (DL) layer supports the transference of data by arranging the bits it receives from the PHY layer into frames and appends a header. This layer also encapsulates higher layer protocols, that is, all the layers less the PHY layer.

12.2.3 Network Layer

The network (NW) layer converts the frame received from the DL layer into packets, appends a header, and forwards them through the network, using IP addresses via routers. IP addresses are analogous to street addresses used by the post office. The NW layer encapsulates higher layers—all except the PHY and DL layers.

12.2.4 Transport Layer

Identified as layer 4, the transport (TP) layer supports two transport protocols: Transport Control Protocol (TCP) and User Datagram Protocol (UDP). TCP supports a reliable, error recovery, and flow control end-to-end packet transmission. User Datagram Protocol (UDP) is a faster but less accurate version of TCP. TCP is stream oriented, while UDP is connectionless. (TCP uses a port number analogous to an apartment number used by the post office to deliver mail.) TP encapsulates application-layer data.

12.2.5 Application Layer

The fifth layer is the application layer, which comprises the application software the user runs and the very information to be conveyed. The application software runs, for example, a Web browser and an e-mail application. Simple Mail Transfer Protocol (SMTP) and Post Office Protocol version 3 (POP3) are for sending and receiving emails, respectively, and they use different port numbers.

12.2.6 A Router

Operating on the IP or NW layer, a router forwards data packets, depending on the forwarding address included within the packets. For this purpose, a router contains an address and forwarding node lookup table. By comparing the forwarding address in the header of the packet to the address entry in the lookup table, we can determine the corresponding next router or node. In other words, at this stage of the NW layer, the packets received are examined for their destination addresses before they are forwarded to the next or final delivery address, node, or router. At least two other routers need to be connected to a router.

12.2.7 A Bridge

In contrast to a router, a bridge works at the data link layer and relays frames according to the link layer header. It is possible for routers and bridges to be interconnected, in which case the bridge is invisible to the network layer. The most common examples of data link layer applications are Ethernet, Point-to-Point Protocol (PPP) and ATM. In Ethernet applications, the blocks of data called frames are appended with hardware destination and source addresses, in addition to other control and error-checking fields.

It is important to emphasize that the physical layer and data link layer operate at hardware levels, while network the layer,and higher layers are strictly software-based.

To shed light onto how the various layers interrelate, it is instructive to view them as one protocol layer encapsulated in the other layer, with the exception of the application layer. Thus:

A = Application protocol + data;
TP = Transport protocol + A;
NW = NW protocol + TP;
DL = DL protocol + NW;
PHY = PHY protocol + DL.

Within the protocol are typically included the header, address, control, and error-checking fields.

Now, the PHY layer encapsulates all the other higher layers ready to ship the data over the PHY layer, the air interface, in our case. Hence:

$$\text{PHY} = \text{PHY protocol} + (\text{DL protocol} + \\ (\text{NW protocol} + (\text{transport protocol} + (\text{application protocol} + \text{data}))))$$

Along the way, if, for instance, a bridge is encountered, the above data will be dismantled up to the DL layer to determine which next hardware address the frame should be forwarded to. Thus:

DL protocol + (NW protocol + (transport protocol +
(application protocol + data))

After the address is identified, the entire set of protocols and data are reassembled before forwarding to the next node. If the next node is a router, the data is stripped to the NW layer in order to look at the next routing address:

NW protocol + (transport protocol + (application protocol + data))

It should be underlined that the data and protocols of the higher layers, in these instances, are always kept intact in the routing process. While the data is in transit, only the necessary headers and control fields are looked at, largely to determine the next forwarding address or to perform error checking. It is only at the final destination, the application layer, that the data is accessed and used as necessary, and the rest is discarded.

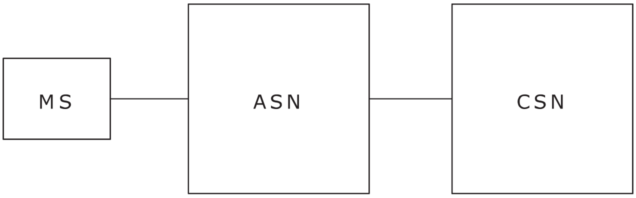
12.3 WiMAX Architecture

Any other IP-based network, including the discussion of the WiMAX network that we are about to embark upon, operates in a fashion similar to what we just discussed. In addition, what makes WiMAX stand out as a convincing technology is the fact that it has added such crucial features as QoS and security at the DL and higher layers.

WiMAX architecture is similar in many ways to a wide-area IP access network [2], such as the universal mobile telecommunication system (UMTS). Whereas a link layer, such as access service network (ASN) is used to connect users to an IP-based application or services and entities such as connectivity service network (CSN), similarly the UMTS terrestrial radio access network (UTRAN) is used as a link between the user equipment (UE) and core network (CN). Please refer to Figures 12.3 and 12.4.

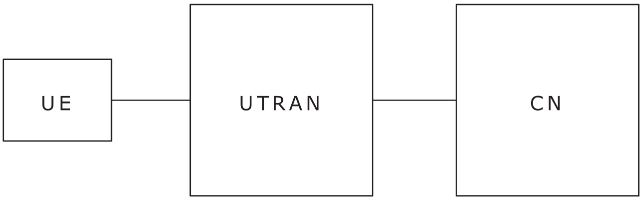
12.3.1 ASN

ASN includes BSs, base station transceivers (BTSs), base station controllers (BSCs), and gateways. ASN also includes the AAA proxy client and dynamic host configuration protocol (DHCP) relay interfacing to the authorization authentication accounting (AAA) server and the address allocation server (AAS), respectively, in the CSN (not shown in the figure, but shown in WiMAX security discussion, in Chapter 13). While connecting the MSs with IP-based services in CSN and comprising the BSs and gateways, ASN assists in radio resources management (RRM) and allocation, mobility, and handover, security,



MS = Mobile Station
ASN = Access Service Network
CSN = Connectivity Service Network

Figure 12.3 WiMAX network.



UE = User Equipment
UTRAN = UMTS Terrestrial Radio Access Network
CN = Core Network

Figure 12.4 UMTS architecture.

paging, and QoS. Refer to Figure 12.5, which shows the interconnection and the main components inside the ASN.

12.3.2 CSN

CSN is comprised of routers, servers, and interfaces to the Internet, the gateway of the ASN and another CSN. The servers within CSN include AAA and AAS. Refer to Figure 12.6, which displays the major components comprising CSN [3].

CSN serves the following purposes:

- Provides connectivity to the Internet and other public networks;
- Provides AAA server;
- Policy management for QoS and security;
- Billing;
- IP address allocation and management to the MSs;

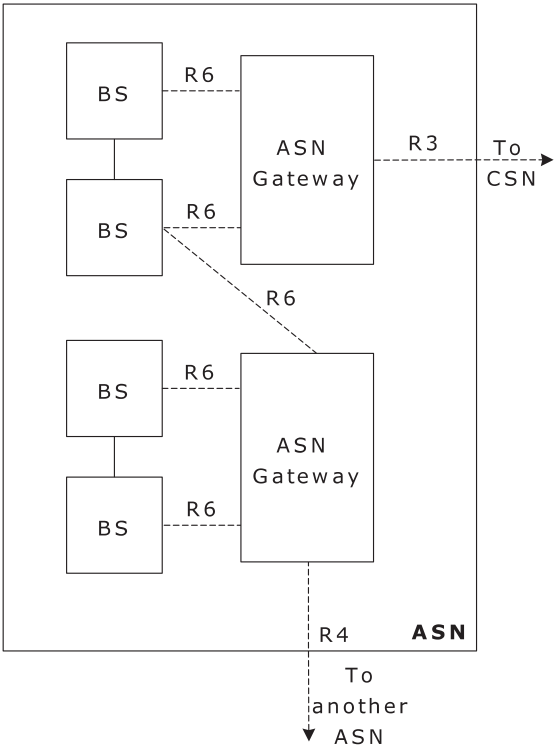


Figure 12.5 ASN components.

- Support inter-CSN roaming and mobility;
- Location management between ASNs;
- Provide interface via gateways to other networks, such as PSTN, 3GPP (GSM) and 3GPP2 (CDMA2000).

The interfaces, R1–R8, are defined as follows:

- R1 = MS←→ASN, air interface between MS and BS within the ASN;
- R2 = MS←→CSN, for authentication, authorization, IP configuration, and mobility management;
- R3 = ASN←→CSN, policy enforcement, and mobility management;
- R4 = ASN←→ASN, mobility within ASN;
- R5 = CSN←→CSN, mobility between network providers;

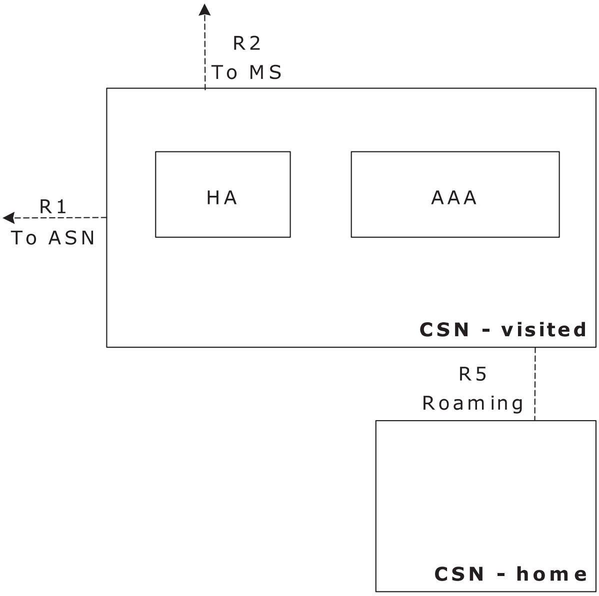


Figure 12.6 CSN interconnection.

- R6 = BS←→ASN-gateway, intra-ASN bearer paths and IP tunnels for mobility events;
- R7 = optional control plane protocol in R6 transactions, facilitates handover;
- R8 = BS←→BS, facilitates handover.

12.3.3 WiMAX Protocol Stack

The crucial parts of WiMAX are the physical and data link layers.

Refer to Figure 12.7. The data link layer is subdivided into three.

The MAC layer as part of the data link layer is an interface between transport and physical layer. Refer to Figure 12.7, depicting the WiMAX protocol stack.

Functionally, the MAC layer takes packets from MAC service data units—MSDUs—and arranges them into MAC protocol data units—MPDUs to be transmitted over the physical layer or air interface. The convergence sublayer is the upper part of the MAC layer interfaces to the IP, network layer, while the privacy sublayer interfaces to the physical layer.

The functional purposes of the three sublayers of the data link layer and the MAC layer are:

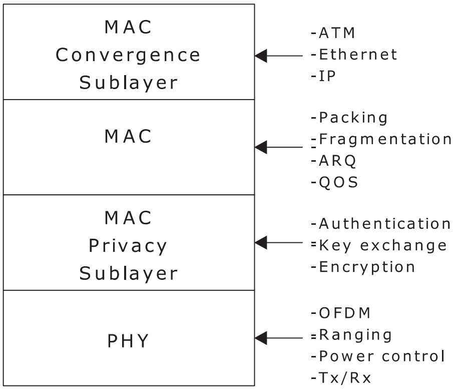


Figure 12.7 WiMAX protocol stack.

- MAC privacy sublayer, where most of the authentication, encryption, and key exchange for traffic encryption is handled;
- MAC sublayer, the sublayer where framing, packing, error handling, and quality of service are supported;
- MAC convergence sublayer, where an upper layer packet can be encapsulated on transmission or can translate upper layer QoS parameters to ones consistent with the MAC layer. These processes are reversed on reception.

12.3.4 MAC and QoS

Wireless local area network (WLAN), or sometimes also known as wireless fidelity (WiFi), is a contention-based access MAC application technology. In other words, a subscriber wanting to send data must first get the attention of the access point (AP) and can experience considerable delay, depending on the number of users asking for the same service at about the *same time*. This delay can unfortunately translate into a reduction of throughput, a measure of the efficiency of passing data through the network.

In contrast to WLAN, the 802.16 MAC uses a scheduling algorithm to assign a slot once a user is admitted for service. Once users gain entry into the network, they are allocated access slots initially by the BS, and they do not compete for attention whenever they need to send data. The time slot assigned to a user may increase or decrease, depending on the demands on the system, but the slot scheduled for that user will not be forfeited. Clearly, this approach is advantageous from a standpoint of optimizing system overload, maintaining bandwidth efficiency, allowing the BS to control the QoS parameter by balancing the time slot among the application needs of the SS's. As an extension of the function just described, the MAC layer supports:

- Scheduling of data;
- QoS setup and maintenance;
- Connection management;
- Handovers;
- Idle and sleep mode of the MS.

The MAC layer, as a connection-oriented layer, is capable of supporting two modes: MS ←→BS (Point-to-multipoint mode) and MS←→MS (mesh mode).

12.3.4.1 Quality of Service (QoS)

QoS design in the MAC layer is one of the attractions of WiMAX technology. This is achieved by using a connection-oriented MAC architecture, where the uplink and downlink connections are controlled by the serving BS. Prior to data transmission, the BS and MS (SS and MS are used interchangeably throughout this book) establish a unidirectional link called a *connection*, which is not physical, but logical. The connection occurs between the MAC layer in the BS and MAC layer in the MS. Upon establishing this link, between the layer peers, each connection is identified by a connection identifier (CID), which serves as a temporary address for the data transmission over the established link. In addition to establishing the connections just described, the WiMAX MAC defines three management connections: (1) basic; (2) primary; and (3) secondary. Finally, WiMAX also defines the concept of a service flow, which is a unidirectional flow of packets with a particular set of QoS parameters. It is identified by a service flow identifier (SFID). There are three types of service flow: (1) provisioned; (2) admitted; and (3) active.

The QoS parameters could include:

- Traffic priority;
- Maximum sustained traffic rate;
- Maximum burst rate;
- Minimum tolerable rate;
- Scheduling type;
- ARQ type;
- Maximum delay;

- Tolerated jitter;
- Service data limit type and size;
- Bandwidth request mechanism to be used;
- Transmission PDU formation rates, and so on.

The BS is responsible for issuing SFID and mapping it to unique CIDs. The service flows can also be mapped to differentiated service-code-point (DiffServ or DSCP), or multiprotocol-label-switching (MPLS) flow labels to enable end-to-end, IP-based QoS. QoS protocols will be discussed in Section 12.3.6.

In order to support a wide variety of applications, WiMAX defines five schedules.

12.3.5 Scheduling

A connection and QoS parameters make up a service flow [2]. There are five different QoS-class scheduling mechanisms defined by the standard:

1. *Unsolicited grant services:* The unsolicited grant services (UGS) are designed to support fixed-size data packets at a constant bit rate. An example of a UGS is VOIP without silence suppression.
2. *Real-time polling service:* The real-time polling service (rtPS) is designed to support real-time services flow, such as MPEG video.
3. *Nonreal-time polling service:* The nonreal-time polling service (nrtPS) is designed to support a delay-tolerant data stream, such as FTP, requiring variable-sized data grants at a minimum guaranteed rate.
4. *Best-effort service:* Best-effort (BE) service is designed for applications, such as Web browsing, that don't required a minimum service level guarantee
5. *Extended real-time variable rate service:* Extended real-time variable rate (ERT-VR) service (also referred to as extended real-time polling service (ErtPS) is designed to support real-time applications, such as VOIP with silence suppression, that have variable data rates, but require guaranteed data rate and delay.

12.3.6 QoS Protocols

Three of the most important QoS protocols are: differentiated service (DiffServ or DS), integrated services (IntServ), and multiprotocol label switching (MPLS) [4].

12.3.6.1 DiffServ/DS

DiffServ or DS, operating on the NW layer, is designed to solve the IP quality-of-service problem. DS takes the IP type-of-service field, renames it the DiffServ byte, and uses it to define the service quality requirement of the IP packet. The DiffServ function is not applied at every router, but rather on a group of them. DS divides the traffic into a small number of classes and treats each class differently, thereby creating the DiffServ code point (DSCP). As an example, when a user sends traffic into a DiffServ network, the same marks each packet with DSCP. The ingress router queues the packets accordingly, while rejecting ones without DSCP. Within the network, the routers queue and schedule delivery by applying the DSCP. A certain class, for example, can get expedited forwarding. The Internet Engineering Task Force (IETF) defines the architecture and semantics of DiffServ.

12.3.6.2 Integrated Service

Using the Resource Reservation Protocol (RSVP), the integrated service (IntServ) provides a QoS guarantee on a per-hop basis throughout the IP network. IntServ, for example, will allow video and sound to reach the recipient without interruption. IntServ is a fine-grained QoS system, while DiffServ is a course-grained control system.

12.3.6.3 MPLS

MPLS adds a label containing specific routing information to each IP packet and allows routers to assign explicit paths to various classes of traffic. MPLS also offers a traffic engineering tool that improves IP forwarding efficiency. Additionally, it also allows the improvement of integration in the IP-over-ATM application. MPLS resides between layer 2 and the IP headers. Thus, in MPLS, network packets are routed based on these labels rather than on the IP headers. Refer to Figure 12.8, which displays the MPLS function.

The ingress router (refer to Figure 12.8) is responsible for inserting the label on each incoming packet, and this label is inserted according to a class called forward equivalency class (FEC). A packet with the same FEC will be routed on predefined paths prior to transmission.

12.3.7 Session Protocol

Session Initiation Protocol (SIP), while not a QoS protocol, runs over IP and is a key in the IP-based telephony. It is powerful in shaping today's and tomorrow's world of telecom. Running over IP and sitting at the application level, SIP is a request-response protocol, similar to the HTTP and SMTP Internet applications and is crucial in setting up, manipulating, and tearing down a session. SIP is an RFC3261 standard from the Internet Engineering Task Force (IETF).

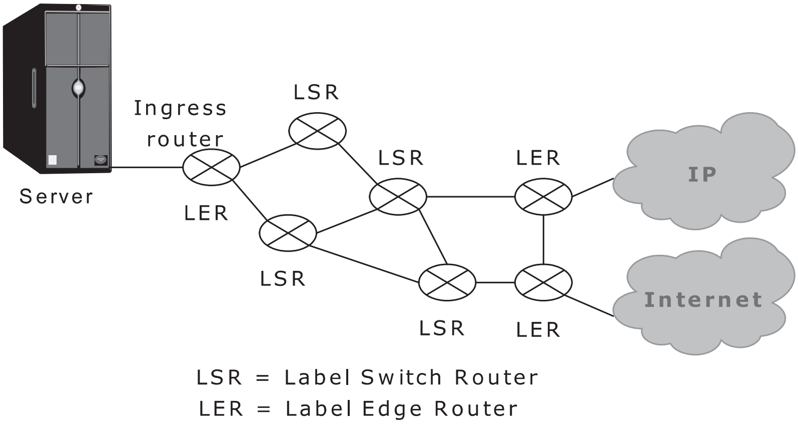


Figure 12.8 MPLS architecture.

Another similar but less versatile and yet and more complex session protocol, along with its other supporting protocols, is H.323. Another less expensive, not as versatile call-control protocol version of SIP and H.323, is the media gateway control protocol or media gateway controller (MGCP/MEGACO). MGCP is the original and MEGACO is the evolved version. Devices that support MGCP/MEGACO also support SIP and H.323.

12.4 WiMAX Frame Structures

WiMAX supports both time division duplex (TDD) and frequency division duplex (FDD) in either OFDM or OFDMA flavors. TDD is more applicable to asymmetric applications, such as the Internet, where the downlink data rate is much higher than the uplink rate. Refer to Figure 12.9, which displays the OFDM frame structure. TDD uses the same frequency for uplink and downlink, separated by time only. Largely deemed obsolete, FDD assigns fixed frequencies for uplink and downlink and is traditionally used for voice-only and two-way radio applications. Voice traffic is predictable, while data traffic is not. TDD provides more spectrum efficiency than FDD, and, hence, is more amenable for data usage.

12.4.1 TDD Virtues

In TDD, uplink and downlink frames or bandwidth can be adjusted (by the service provider) according to demand. Uplink power is considerably lower than downlink power, on account of SS's limited power and antenna gain.

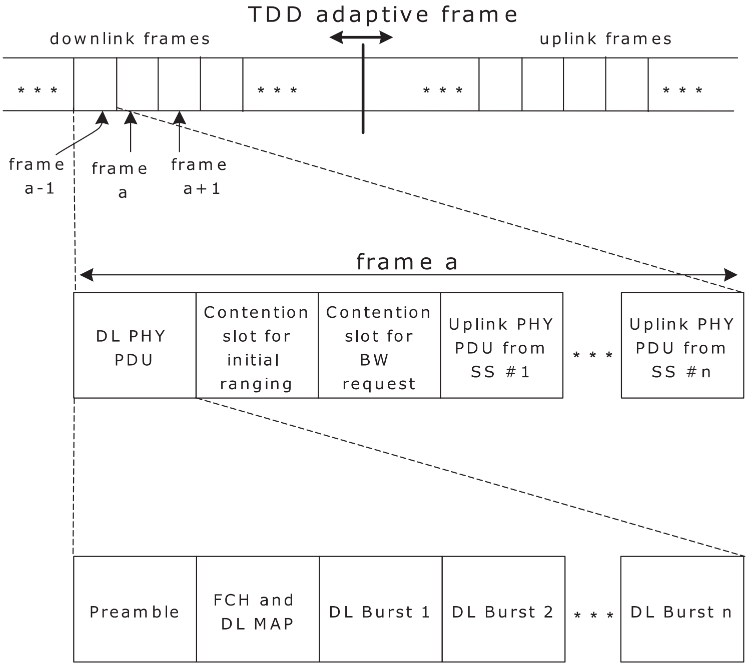


Figure 12.9 Frame and data bursts structure.

The ratio between uplink and downlink can be configured 2:1, 3:1, 4:1, and so on, to give the downlink maximum capacity. A ratio of 1:1 gives the uplink the maximum capacity. While limited in its power, with more flexible bandwidth and more frames available for uplink, the SS enjoys more capacity. Other WiMAX benefits, such as MIMO and smart antennas, should assist the SS with its power quest as well.

12.4.2 OFDMA TDD Frame Structure

In Figure 12.10, on the OFDMA TDD frame structure where subcarriers (vertically) and symbols (horizontally) are shown, it can be observed that 48 symbols with durations of 102 μs are fitted into one 5-ms frame. In the figure are shown:

- Uplink and downlink maps (UL MAP and DL MAP) specify or describe uplink and downlink bursts and burst times, respectively.
- A downlink burst (DL burst), is the transmission of MAC PDU.
- An uplink burst (UL burst) is the reception of a MAC PDU by a single user.
- Initial ranging has SS send a ranging message.

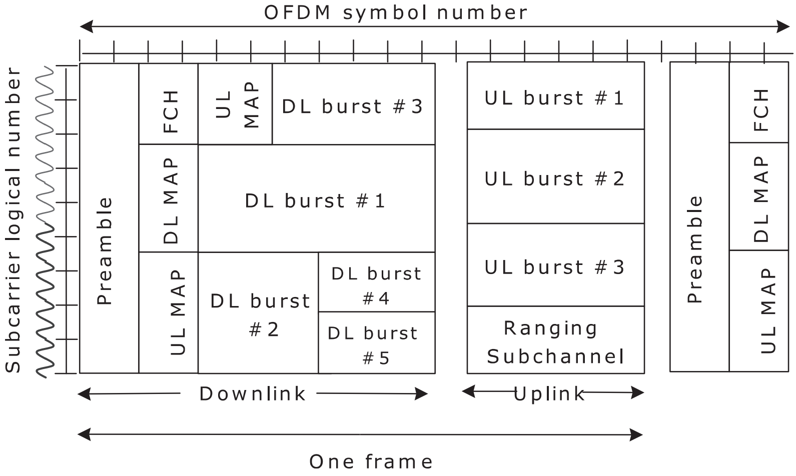


Figure 12.10 Subcarrier, symbol, and frame structure.

- A bandwidth request has SS send a bandwidth request message.
- A preamble is used for frame synchronization and equalization.
- A frame control header (FCH) (one symbol long) contains the location and burst profile of the first DL burst.

In general, the downlink data transmission takes place roughly in the following way. The user-generated data bits at the transmitter are coded typically using a convolutional encoder, with an applicable code rate. Encoded data bits are then interleaved, followed by mapping to modulated symbols. The modulated symbols modulate the subcarriers to form DL bursts. At this point Preamble and DL-MAP are added to form the DL subframe. Various parameters help determine the number and the insertion location of the bursts within the subframes. The IFFT transforms the OFDMA symbols to create OFDMA waveforms, and then the cyclic prefixes are inserted. The full usage of subchannel (FUSC) and/or the partial usage of subchannel (PUSC) are then implemented. Finally, parallel to serial, followed by digital-to-analog, conversions take place, followed by RF modulation to transmit the signal over the air via the antenna.

12.4.3 Subchannels, Slots, Frames, and Permutation

A subchannel, as underscored before, is a group of subcarriers. The permutation mode determines how the subcarriers making up subchannels are distributed.

Subchannels are either adjacent to each other or noncontiguously distributed across frequency band depending on the subcarrier permutation mode:

1. *Diversity (or distributed) permutation:* Where the subcarriers are distributed pseudo-randomly. The main advantages of distributed permutations are frequency diversity and minimizing mutual interference between cells. Examples of this are FUSC and PUSC.
2. *Contiguous (or adjacent) permutations:* Where subcarriers are adjacent. An example of this is the AMC mode. Channel estimation is easier, as the subcarriers are adjacent.

A slot is the smallest unit of data allocation in 802.16 and has both time and subchannel dimensions. Slot definition varies according to uplink and downlink; for FUSC and PUSC, distribute diversity permutation and adjacent subcarrier permutation. The MAC layer allocates the time/frequency resources to users in units of slots. The size of a slot depends on the subcarriers permutation mode:

- *FUSC:* Each slot is 48 subcarriers by one OFDM symbol.
- *Downlink PUSC ("cluster"):* Each slot is 24 data subcarriers by two OFDMA symbols (again a total of 48 tone-symbol).
- *Uplink PUSC and TUSC ("tile"):* Each slot is 16 data subcarriers by three OFDMA symbols (a total of 48 tone-symbol).
- *AMC ("bin"):* 6 adjacent bins over 1 symbol; 3 adjacent bins over 2 symbols; 2 adjacent bins over 3 symbols; 1 bin over 6 symbols. (Each bin has 9 adjacent tones: 8 data and 1 pilot, 48 tone-symbols in each case.)

In Figure 12.11, it can be seen that a minimum assignable resource, 1 slot = 48 subcarriers × 1 symbol, is appropriated to each SS. For example, at the bottom of the figure, SS1, subscriber station 1, is assigned four slots all of the time; SS2 is assigned two slots half of the time and so is SS3. SS4 is assigned four slots all of the time.

12.5 Introduction to VoIP

VoIP is one of the applications of WiMAX technology. The fundamental VoIP protocol is shown in Figure 12.11. WiMAX VoIP will depart somewhat from

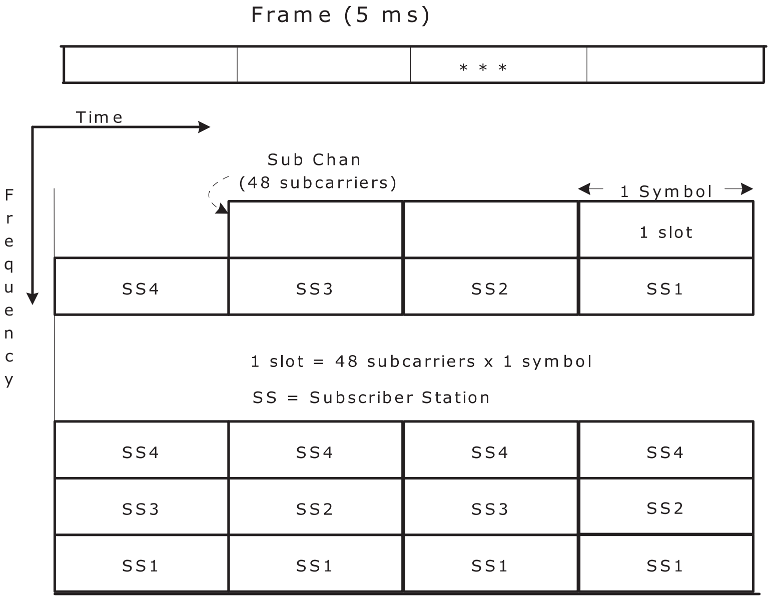


Figure 12.11 Slots and frames.

this figure. The data link layer in this case comprises Point-to-Point Protocol (PPP), link access procedure for modems (LAPM), and HDLC [5].

The digitized voice is the VoIP traffic of the user at the application level. The voice traffic is carried by the Real Time Protocol (RTP). RTP is designed to carry voice and video data and runs on UDP as shown in Figure 12.12. RTP contains time stamp and synchronization information in the header and the sequences of the traffic.

UDP, which is connectionless, is used in carrying real-time data, such as speech in this application. Because it is meant to be more efficient, it is not as reliable as TCP. TCP, the connection-based application, is reliable and allows retransmission data or resends a lost packet. However, that is not a luxury in real-time voice application.

As shown in Figure 12.12, IP is the protocol at the network level. At this layer, packets containing a forwarding address go through the network of routers to the final destination. It is also possible to have an ATM backbone in which case data will only be examined on the layer 2 level. This effectively reduces the processing time. See Figure 12.13 showing the ATM backbone protocol stack.

The PPP is mostly used for dial-up connection to the Internet and is useful in negotiating which type of network layer protocol should be used during the session and authentication procedures. More importantly, PPP is capable of subsuming a network layer datagram from the higher layer over the serial link, a

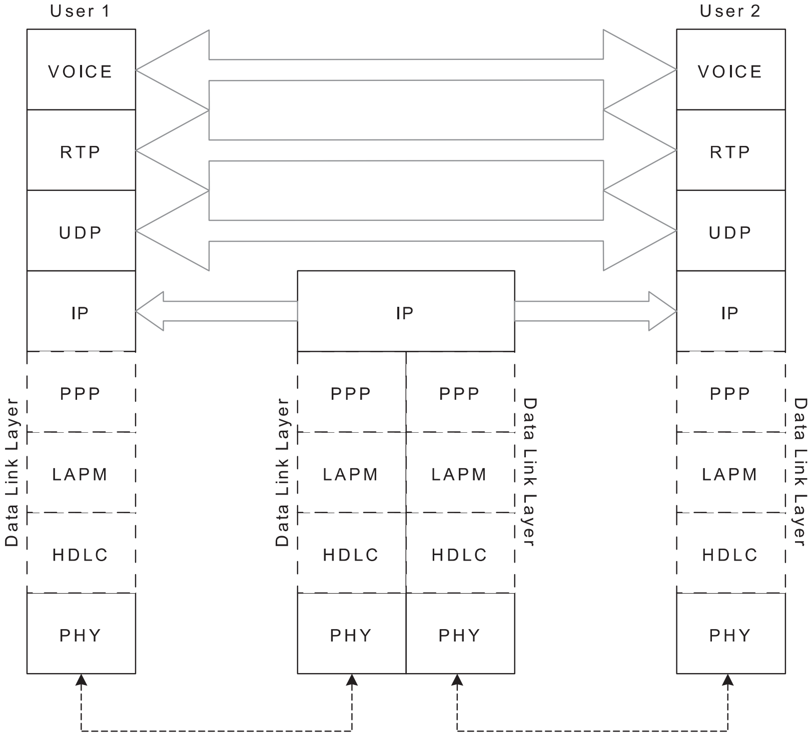


Figure 12.12 VoIP protocol stack.

typical link for the dial-up application. PPP, assisted by HDLC, performs error checking and bit-staffing. LAPM, assisted by HDLC, performs the data link handshaking after physical layer and insures proper sequencing of traffic. If the frames are out of order, it notifies the model of the problem. HDLC performs the layer 2 framing error checking and calculation and frame assembling and disassembling functions.

The physical layer in this dial-up network is an analog signal over cable using a high-speed modem such as V.90. ISDN users are supported by V.100.

12.6 VoIP over WiMAX

As a nascent technology and with strong IP-based wireless broadband capability, it is believed VoIP and WiMAX will be an ideal match. At the time of this writing, VoIP over WiMAX is being tested in various locations and by different service providers around the world. Preliminary test results show promising results, with a lot of room left for improvement. Clearly, availability of spectrum,

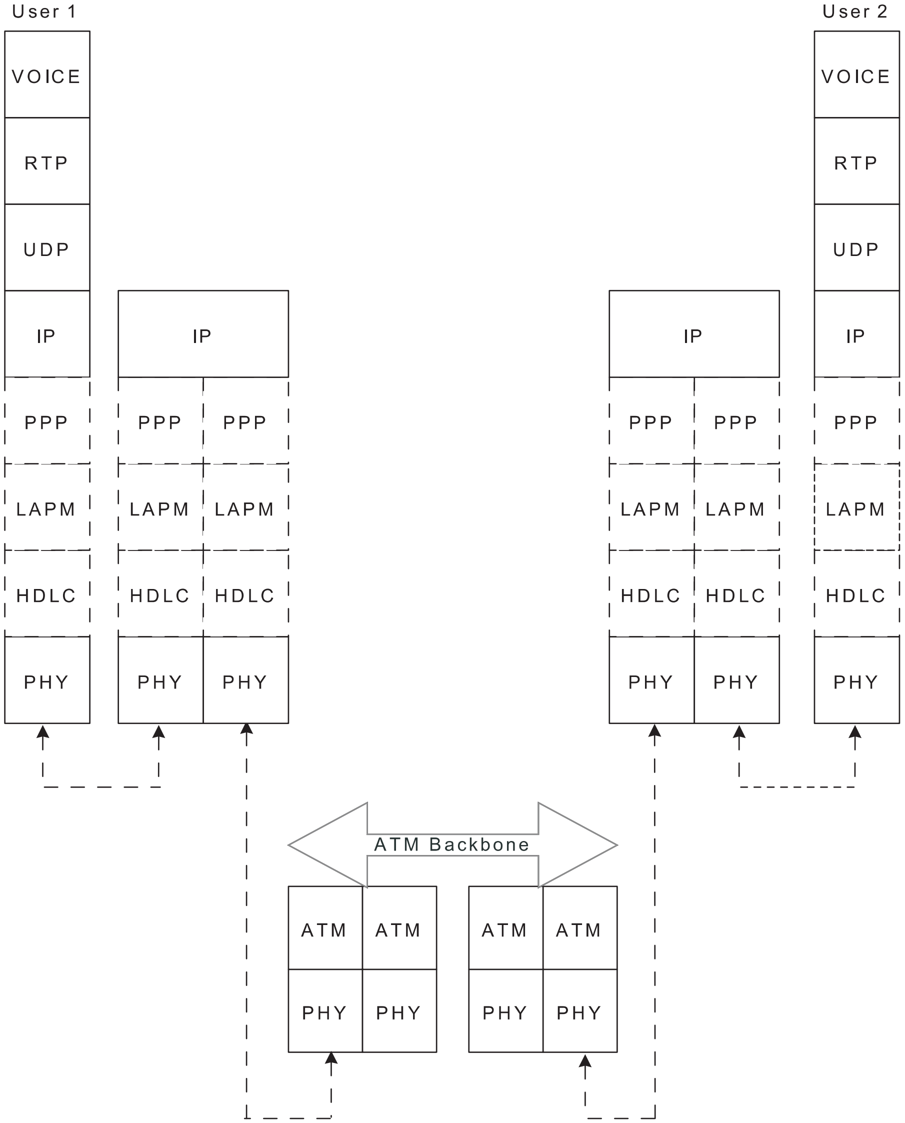


Figure 12.13 VoIP with ATM.

business decisions, and market opportunity, among other reasons, drive how soon this side of the technology will be deployed more widely in the United States.

The greatest appeal of WiMAX as described in this text is the fact that it is a wireless broadband in its variations: fixed, portable, and mobile. Currently more people are using less plain old telephone (POT). The major WiMAX players in the United States today are Clearwire and Sprint-Nextel in terms of network providers. From the equipment manufacturing side, Nokia, Intel, Motorola, Alvarion, and Airspan have gained visibility with real working models of handsets, modems, BSs, and network infrastructures with good products already supplying WiMAX networks around the world.

In the United States, WiMAX is not yet ubiquitous. The first and only mobile WiMAX was deployed in Baltimore, Maryland, in September 2008. Many believe that once the appeal is understood by the public an avalanche of prospective customers will flock for the services, which will spur further buildout.

Users, as always, want one billing for all services. The increase of use of WiFi hotspots in neighborhood cafés is another indicator that the trend toward wireless broadband usage is clearly there. Hence, once WiMAX becomes one-stop, all-service technology, cable, DSL Internet, and phone services and then WiFi hotspots, landline phone services, and eventually cellular/PCS services in some cases will be vulnerable to losing market share. A few service providers, seeing the writing on the wall, are beginning to reposition themselves. In the meantime, outside the United States, places like Brazil, with a population approaching 200 million, have embraced the WiMAX technology and derived many benefits. Historically, countries such as Brazil are more accepting of new technology with appeal to rural and developing areas of the population segment. WiMAX is very applicable and well suited and cheaper to deploy in remote areas where there is no landline phone, cable, or even electricity, especially since photovoltaic microsolar panels are now a reality.

References

[1] Solomon, J., *Mobile IP: The Internet Unplugged,* Upper Saddle River, NJ: Prentice-Hall, 1998.

[2] Annelie, D., and Michael, C. L., “WiMAX—A Study of Mobility and a MAC-Layer Implementation in GloMoSim,” Master’s thesis in computer science, Umea University, Umea, Sweden, April 6, 2006.

[3] Ergen, M., “The Access Service Network in WiMAX: The Role of ASN-GW,” W. Chorus, Inc., 2007.

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13

WiMAX Security

13.1 Introduction

Security is vital to communication. It is even more decisive when the communication is wireless and in a mobile environment.

It goes without saying, from a security vantage point, that WiMAX has to outperform the rivals it is looking to replace. Some of the technologies to which WiMAX is posing immediate challenges are WiFi, broadband cable, and digital subscriber line (DSL). (DSL uses the telephone lines to superimpose high-speed data.) It can be argued, though, that WiMAX does not always have to compete with these technologies, and that, at times, it can work in conjunction with them to complement them in an overlay or contiguous fashion. Be that as it may, the burden is still on WiMAX, as a newcomer, to be a convincing technology from a security perspective.

We know how the security gap in WiFi limited its wider uses. For example, a rogue BS can steal a WiFi access point (AP) ID and construct a message legitimate-looking enough to cause it to give up private information. Thankfully, those critical security issues uncovered in WiFi have been addressed in the WiMAX standard. When WiMAX is deployed more widely than it is today, no doubt it is possible to identify additional security issues hitherto unknown. The WiMAX standard has reasonable security foundations to build on. In the final analysis, the burden is on the vendors and service providers to go above and beyond the foundational security measures spelled out in the standard to guarantee a more secure network.

13.2 Documents Related to the WiMAX Security Standard

Data-link security sublayer functions are defined by IEEE802-16e-2005 standards for the WiMAX air interface, while the network aspects of security are in accordance with the WiMAX forum network reference model (NRM). RFC 2903 contains the framework that specifies the protocol and procedure for authentication, authorization, and accounting associated with the user.

The WiMAX technology standard specifies:

- Private key management (PKMv2) Extensible Authentication Protocol (EAP)-based standard between the device and the BS;
- Over-the-air advanced encryption standard (AES)-based encryption for subscriber traffic;
- Authentication, authorization, and accounting (AAA) protocols for device authentication. AAA protocol is for network authentication and when the intelligence is not available in the BS.

13.3 Fundamental Data Encryption Techniques

Since not all algorithms are supported by all users, various types of encryption methodologies are negotiated prior to the commencement of data transactions. For data encryption a number of standards and algorithms are used:

1. *Data encryption standards (DES):* DES is a symmetric encryption in that both the MS and BS use the same key. DES uses a 56-bit encryption key.
2. *3DES:* 3DES uses three times the DES number of bits: $3 \cdot 56 = 168$-bit encryption key.
3. *Advanced encryption standard (AES):* AES is also symmetric encryption, but it uses different length keys: 128, 192, or 256 keys in the MS and BS.
4. *Ron Rivest, Adi Shamir, and Leonard Adleman (RSA):* RSA involves public and private keys—the message is encrypted using the public key, but decrypted using the private or secret key. RSA uses 1,024–2,048 bits.

13.4 Typical Security Concerns

What are some of the potential security problems encountered in wireless broadband networks such as WiMAX?

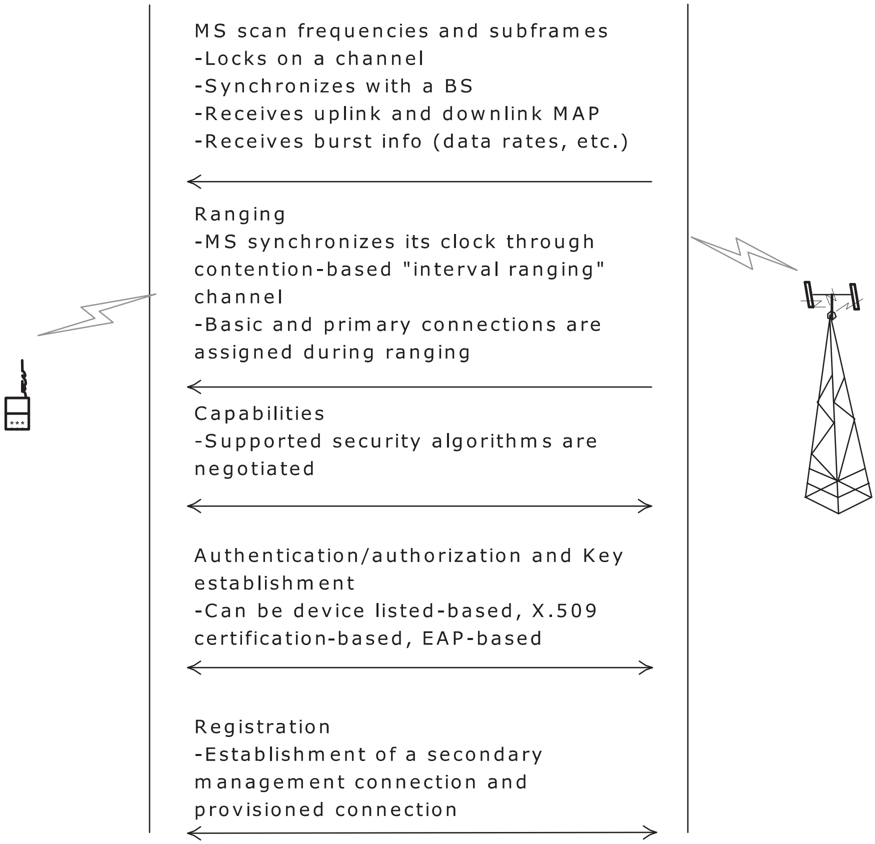
1. *Denial-of-service (DOS) attack:* A DOS attack is a case in which the MS is asked to authenticate itself or process data many times over, thereby keeping it inundated with a lot of requests to the point when time expires and, consequently, service is denied.
2. *Eavesdropping (especially for ViIP):* Eavesdropping is a case in which an attacker is simply listening to messages.
3. *IP-address spoofing:* IP-address spoofing is a case of stealing IP addresses in order to impersonate legitimate entities.
4. *Rogue BS:* Rogue BS is a case in which an illegitimate BS steals the ID of a legitimate BS in order to then communicate with a MS to extract private information.
5. *Man-in-the-middle (MIM) attack:* A MIM attack is a case in which an attacker fools person A into thinking that the attacker is person B, and fools person B into thinking that the attacker is person A, in order to gain access to messages between person A and person B.

13.5 Accessing Service

As a mobile station (MS) (interchangeably referred to as a subscriber station (SS)) proceeds to enter the WiMAX network, as shown in Figure 13.1, it scans and locks on the applicable channel, acquires bit and frame synchronization, performs ranging to adjust suitable transmission power, sets up management connections, performs authentication, and acquires IP connectivity through dynamic host configuration protocol (DHCP). Refer to Figure 13.1, depicting the MS/SS network entry steps.

Before the MS completes its access of the network, and traffic data exchanges begin, the following steps must take place to guarantee a secure communication [1]:

1. *Authentication*—to make sure the SS (or MS) is a legitimate user and is permitted to access the network;
2. *Authorization*—to make sure the MS can access the services it requests of the network and determine the types and parameters associated with those services;



MS scan frequencies and subframes
-Locks on a channel
-Synchronizes with a BS
-Receives uplink and downlink MAP
-Receives burst info (data rates, etc.)

Ranging
-MS synchronizes its clock through contention-based "interval ranging" channel
-Basic and primary connections are assigned during ranging

Capabilities
-Supported security algorithms are negotiated

Authentication/authorization and Key establishment
-Can be device listed-based, X.509 certification-based, EAP-based

Registration
-Establishment of a secondary management connection and provisioned connection

Figure 13.1 MS network entry process.

3. *Accounting*—to make sure resource usages are metered and recorded, depending on the type of service, for example, offline, online, prepaid, or postpaid.

13.6 Authentication

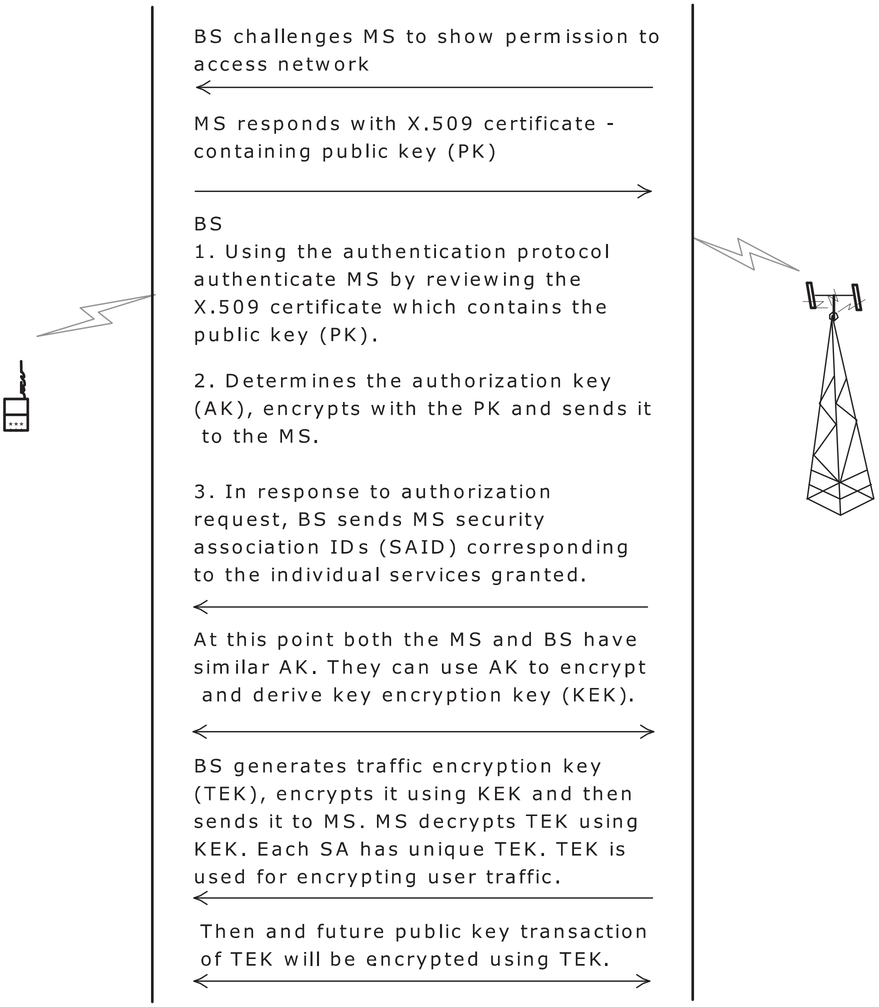
Authentication is comprised not only of verification of the user, but also of key definition using the public key interchange (PKI).

During authentication of MS by BS, MS presents an X.509 certificate as a fulfillment of the requirement for authentication. BS, in response, sends to MS an authentication key (AK) encrypted with a public key (PK). The MS is then able to derive the key exchange key (KEK) from AK.

There are three types of authentications, according to IEEE 802.16e-2005:

1. RSA-based (X.509 authentication with RSA encryption);
2. Extensible authentication protocol (EAP);
3. RSA and EAP authentication.

Authentication, authorization, and data encryption mainly target the data-link layer. Please see Figure 13.2, which summarizes the authentication and authorization steps.



BS challenges MS to show permission to access network

MS responds with X.509 certificate - containing public key (PK)

BS

1. Using the authentication protocol authenticate MS by reviewing the X.509 certificate which contains the public key (PK).

2. Determines the authorization key (AK), encrypts with the PK and sends it to the MS.

3. In response to authorization request, BS sends MS security association IDs (SAID) corresponding to the individual services granted.

At this point both the MS and BS have similar AK. They can use AK to encrypt and derive key encryption key (KEK).

BS generates traffic encryption key (TEK), encrypts it using KEK and then sends it to MS. MS decrypts TEK using KEK. Each SA has unique TEK. TEK is used for encrypting user traffic.

Then and future public key transaction of TEK will be encrypted using TEK.

Figure 13.2 Authentication and authorization.

13.7 Authorization

Authorization is a phase that takes place right after, or, depending on the point of view, at the same time as the authentication process.

In order to be granted service, the MS requests an authentication key (AK) and security association (SA) identity (SAID). SA is security information shared between the MS and the BS specific to the services the MS is authorized to be granted. For this request, the MS presents an X.509 certificate and other encryption algorithms and IDs. In response, the BS, after consulting the AAA server, replies to the MS with an AK encrypted with PK, a lifetime key, and SA.

Since the authorization has to be carried in the AAA server, the BS has to relay the message to the server for this purpose and back to the MS with an AK encrypted with PK, a lifetime key, and a (SA).

It bears emphasizing that authentication and authorization take place almost at the same time. For example, the presentation of X.509 by the MS can be viewed as a partial fulfillment of both the authentication and authorization requirements. It should also be pointed out that even after authentication is carried out and authorization is granted, the encryption and key establishments need to be updated periodically in order to continue to ensure secured communication.

13.8 The Network Side of Security

Extensible Authentication Protocol (EAP) is used in the transaction between the MS and the BS for the purpose of authentication, especially when the intelligence resides within the BS. EAP runs over the MAC and PHY layers, utilizing the PKMv2 protocol. Otherwise, in all other cases, the BS has to relay the authentication request to the AAA server in the core service network (CSN) via the R3 interface. Refer to Figure 13.3, which shows the WiMAX AAA Architecture. Again, EAP is used for the authentication task running over remote authentication dial-in user service (RADIUS). RADIUS, working on network layer, is also used for authorization and accounting tasks. The DHCP client shown in the MS is for the purpose of acquiring the IP address via R1 to the ASN and via R3 to CSN.

When the MS is in motion, in handover, between two ASNs or between two CSNs, while roaming, during paging, and so on, messages, IP addresses and authentications are handled in different ways, the discussion of which is outside the scope of this book.

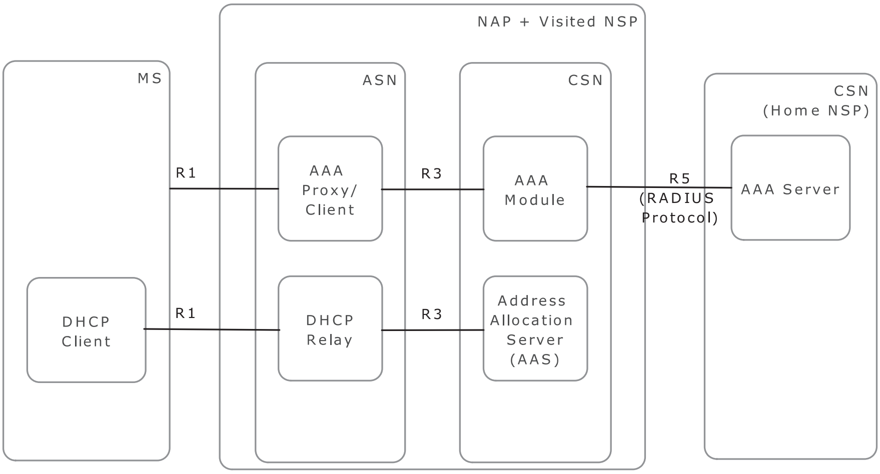


Figure 13.3 WiMAX AAA architecture.

13.9 Security Characterization in WiMAX

Since WiMAX is physical and MAC is a (data-link) layers standard, the security concerns mainly address those two layers.

WiMAX/802.16 is structured into two main layers, MAC and physical. Figure 13.4 shows the protocol stack of WiMAX as relates to security.

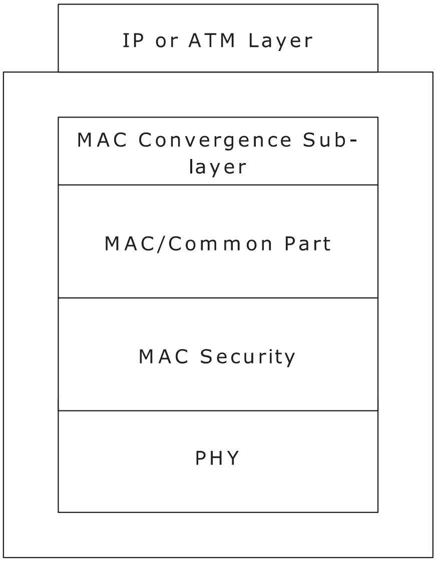


Figure 13.4 WiMAX protocol and security.

The convergence sublayer:

- Adapts the higher layer (for example, IP packets or ATM cells) to the MAC service data unit (SDU). It also does the reverse, adapting the SDU to the higher layers.
- After the adaptation to the SDU, it allots the data according to the connections applicable.

The security sublayer:

- Addresses authentication;
- Establishes key and encryption;
- Exchanges the MAC packet data unit (PDU) with the PHY layer.

The MAC/common part:

- Manages bandwidths;
- Exchanges SDU with the convergence layer;
- Constructs PDUs;
- Establishes connections.

PHY:

- Physical-layer frames are mapped to MAC PDUs and vice versa, for reception and transmission.

13.10 Security Challenges

13.10.1 PHY Layer Protection

The physical layer has to contend with jamming attacks, both deliberate and unintentional. With an eye toward complete disruptions of communication, jamming, for the most part, is a purposeful use of radio frequency to cause severe interference [2].

Jamming can be classified into two categories:

- Interference targeting the entire bandwidth;
- Interference targeting specific frames or time slots.

Physical-layer interferences, especially those indiscriminately targeting the entire bandwidth, including their mitigations, are discussed in Chapter 5.

Refer to Figure 13.5, which shows the physical layer made up of control and data.

13.11 MAC Layer Protection

The MAC header and control/header are not encrypted and are left optional for the service provider. When EAP optionally authenticates the MAC header, it is called hash message authentication code (HMAC) [2].

The part of the payload of the MAC PDU that is user traffic (transport connection) and assigned SAs for uplink and downlink are encrypted using DES or AES.

Management messages of user traffic are not encrypted and are subject to eavesdropping. Refer to Figure 13.6, which depicts the MAC layer PDU.

Typically, the BS unilaterally authenticates the MS. Wherever there is only one-way authentication, as well as other similar authentication deficiencies, the MS can be, in theory, a subject of rogue BS attack, similar to WiFi. However, the offending BS, in the case of TDMA access-based WiMAX, would have to transmit in the same time slot as the legitimate BS is transmitting and do so at much higher power than the legitimate BS—a tall order.

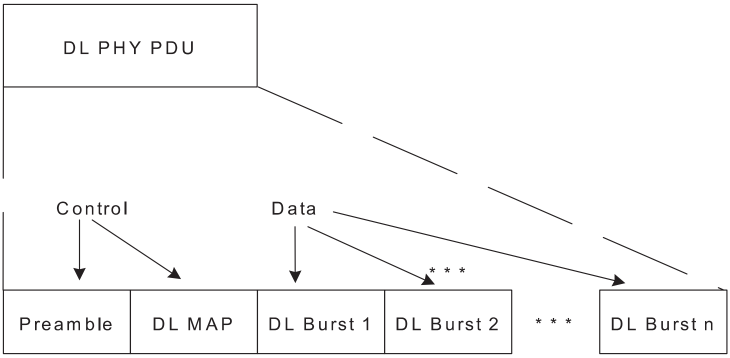


Figure 13.5 Downlink control and data of physical layer.

Figure 13.6 MAC packet data unit.

There are basically three types of authentication and authorization:

- Device-list-based;
- X-509 certificate-based;
- EAP-based.

Device-list-based authentication and authorization are generally used for blocking stolen handsets.

X-509 certificate-based authentication and authorization is for authenticating that the MS. Mutual authentication, a more secured implementation approach, can be supported using EAP-based authentication.

References

[1] Airspan Networks, “Mobile WiMAX Security,” 2007.

[2] Barbeau, M., “WiMAX/802.16 Threat Analysis,” Carleton University, Ontario, Canada, 2005.

Part V
Comparable Systems

14

WiFi Radio Planning

14.1 Overview

WiFi, a cheap, convenient, versatile wireless technology, available now in almost all personal computers, has revolutionized wireless access to the Internet and even, as of late, voice communication through the Internet (VoIP), more profoundly than any other technology in recent memory. From multiple-room, wireless, Internet connections inside residences, to hotspots in favorite cafes, WiFi technology has proved its worth.

Because of these outstanding versatilities, technologists have wanted to push WiFi's application envelopes. While its high data rates and license-exempt spectrums did not disappoint, other shortcomings, such as limited ranges, interference vulnerabilities, and security gaps limited its wider uses. Citywide and seamless deployments proved cost prohibitive.

As a matter of public record, recently, municipality-supported plans to build WiFi networks (also known as Muni WiFi) in several major cities in the United States collapsed on account of exorbitant deployment costs, effectively ending an effort to transform WiFi from a limited-range technology into a city-spanning technology.

It is also a matter of public record that the Homeland Security–sponsored electronic fence along the Mexican border hit several snags when the 802.11a deployment was knocked out of service by outside interference sources using the same spectrum.

It is an established fact that WiFi has huge security issues. Due to its inherent access scheme and authorization and authentication techniques, a rogue access point (AP) posing as a legitimate one can easily construct a legitimate-looking message, causing a subscriber to give up private data.

In this chapter we discuss the fundamentals and design methodologies of WiFi technology, and in so doing, draw contrast to WiMAX.

Among others, spectrum definitions, link budgets, and the range and capacity of WiFi will be examined.

14.2 Channel and Band Assignments

Wireless fidelity (WiFi) and wireless local area network (WLAN) are the generic names for the IEEE 802.11 specification, including, but not limited to, 802.11b, 802.11g, 802.11a, 802.11n, and 802.11-legacy (a legacy specification).

The major specifications are summarized in Table 14.1.

802.11-legacy (a legacy specification) is the specification that led to 802.11b. Most of our discussion will be on standards 802.11a, b, and g. While the 802.11 standard completion was in the works for November 2009, there have been migration efforts by some vendors based on the Draft 2 proposal.

Legacy IEEE 802.11 DSSS systems employ differential binary phase-shift keying (DBPSK) and differential quadrature phase-shift keying (DQPSK) modulation techniques for delivering data packets at data rates of 1 and 2 Mbps.

802.11g has an advantage over 802.11b in terms of higher data rates and the capability to mitigate interference. 802.11g is also capable of interoperating with 802.11b-based devices.

The spectrum of WiFi is as shown in Figure 14.1.

Table 14.1
WiFi Standards

Standard	Release Date	Modulation	Operating Frequency (GHz)	Data Rate (Mbps)		Range (m)
				Typical	Max	
802.11n	November 2009	OFDM	5 and 2.4	74	300	70
802.11g	June 2003	OFDM	2.4	19	54	38
802.11b	October 1999	DSSS	2.4	4.5	11	35
802.11a	October 1999	OFDM	5	23	54	35
802.11-Legacy	1997	DSSS	2.4	1	2	20

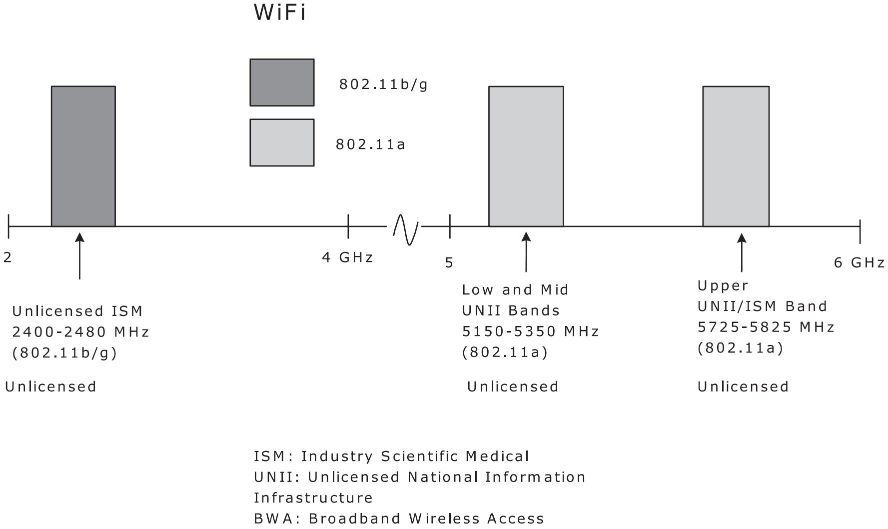


Figure 14.1 WiFi assigned spectrum.

In the United States, 802.11b/g radios use mainly three nonoverlapping channels out of eleven 20-MHz channels. The three nonoverlapping channels are 1, 6, and 11. Refer to Figure 14.2.

802.11a alleviates the frequency crowding in 2.4 GHz and operates in the 5-GHz range. At the same time, it delivers high data rates, while maintaining about the same range as 802.11b. 802.11a radios use one of 12 nonoverlapping 20-MHz channels in the 5-GHz unlicensed national information infrastructure (UNII) band.

The new 802.11n products can use 20- or 40-MHz wide channels in either the ISM or UNII band. The wider bandwidth explains the much higher data rates, as compared to the other specifications.

14.3 FCC Power Requirements

For the sake of reference, we reprint here a portion of the FCC part 15.247 statements governing transmit powers and antenna gains.

> TITLE 47—TELECOMMUNICATION
> CHAPTER I—FEDERAL COMMUNICATIONS COMMISSION
> PART 15—RADIO FREQUENCY DEVICES—Table of Contents
>
> Subpart C—Intentional Radiators

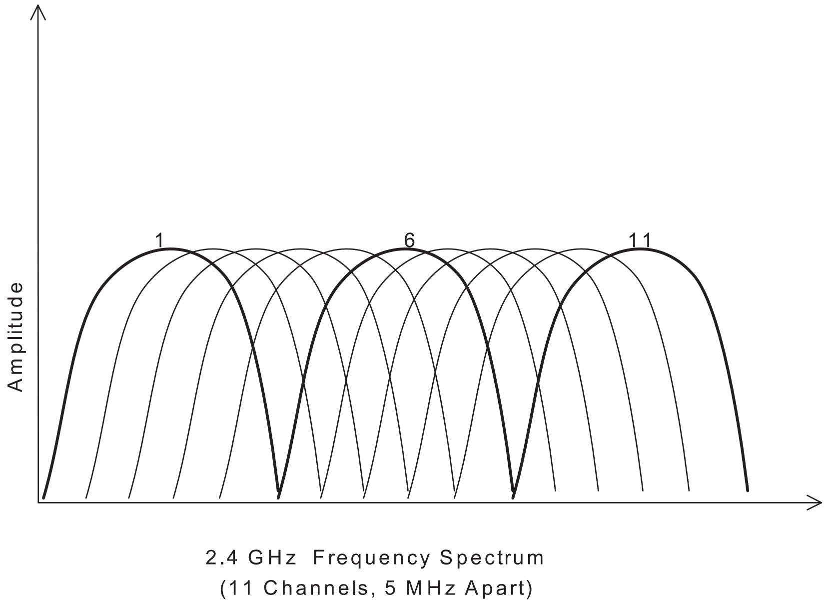


Figure 14.2 Overlapping and nonoverlapping channels. (*From:* [1]. © 2008 J. Geier. Reprinted with permission.)

Sec. 15.247 Operation within the bands 902–928 MHz, 2,400–2,483.5 MHz, and 5,725–5,850 MHz

(b) The maximum peak output power of the transmitter shall not exceed 1 Watt. If transmitting antennas of directional gain greater than 6 dBi are used, the power shall be reduced by the amount in dB that the directional gain of the antenna exceeds 6 dBi.

(3) Except as shown below, if transmitting antennas of directional gain greater than 6 dBi are used the peak output power from the intentional radiator shall be reduced below the above stated values by the amount in dB that the directional gain of the antenna exceeds 6 dBi.

(i) Systems operating in the 2400–2483.5 MHz band that are used exclusively for fixed, point-to-point operations may employ transmitting antennas with directional gain greater than 6 dBi provided the maximum peak output power of the intentional radiator is reduced by 1 dB for every 3 dB that the directional gain of the antenna exceeds 6 dBi.

(ii) Systems operating in the 5725–5850 MHz band that are used exclusively for fixed, point-to-point operations may employ transmitting antennas with directional gain greater than 6 dBi without any corresponding reduction in transmitter peak output power.

(iii) Fixed, point-to-point operation, as used in paragraphs (b)(3)(i) and (b)(3)(ii) of this section, excludes the use of point-to-multipoint systems, omnidirectional applications, and multiple co-located intentional radiators transmitting the same information. The operator of the spread spectrum intentional radiator or, if the equipment is professionally installed, the installer is responsible for ensuring that the system is used exclusively for fixed, point-to-point operations. The instruction manual furnished with the intentional radiator shall contain language in the installation instructions informing the operator and the installer of this responsibility.

14.4 Link Budget

A simple link budget is given in Table 14.2 to illustrate the calculation of the various losses and fade margins.

From the result in Table 14.2, it can be seen that for a 6-km range, we have a 15-dB link budget to play with. The margin available can be used for any other allowances not included in Table 14.2, for testing purposes, where the losses can be absorbed in the devices under testing. In Section 14.5, we will see how well we would do if we were to stretch the path losses and still stay with the FCC guidelines and power requirements.

Table 14.2
WiFi Link Budget for Outdoor, 2.4-GHz Range

Parameters	**Values for Outdoor Downlink**
Transmit power—P_{tx} (dBm)	30
Tx cable loss—L_{txc}(dB)	3
Tx antenna gain—G_{txant} (dBi)	6
Range desired—d (km)	6
Frequency—f (MHz)	2,400
Free Space Path Loss—*Lfspl*, wrt Range (dB)	$= 32.45 + 20\log f + 20\log d$ $= 32.45 + 20\log 2{,}400 + 20\log 6 = 116$
Rx antenna gain—G_{rxant}(dBi)	10
Rx cable loss (dB)—L_{rxc}	3
Rx noise figure—*NF* (dB)	7
Bandwidth—BW (dB)	5 MHz (67 dB)
Signal-to-noise ratio (SNR) (dB)	15
Rx Sensitivity – RX_{sens}(dBm)	$= –174$ dBm/Hz $+ NF + 10\log\text{BW} + \text{SNR}$ $= –174 + 7 + \log(5 \times 10^6) + 15 = –85$ dBm
Fade Margin (dB)	$= P_{tx}\text{-}L_{txc} + G_{txant} – L_{fspl} – L_{rxc} + G_{rxant} – RX_{sens}$ $= 30 – 3 + 6 – 116 – 3 + 10 – (–85) = 15$

14.5 The Biggest Recorded Range

It was brought to our attention that engineers in Venezuela tried to stretch the limit of the WiFi range, while keeping within the FCC power and antenna constraints described in the Section 14.4.

To reiterate by paraphrasing the FCC rule pertaining to point-to-multipoint (PtMP): Service providers who would like to communicate or operate under this portion are only allowed up to 30 dBm or 1 watt of transmitter power output with a 6-dBi antenna or 36-dBm or 4 watts of effective radiated power over an equivocally isotropic, radiated power antenna. The transmitter power output needs to be reduced 1dB for every dB of antenna gain over 6 dBi.

We shall verify the parameters on a Web site by Glenn Fleishman. The claim was that the range was 270 km. Because of the high antenna gain used, 30 dBi, the transmit power was reduced from 1W (30 dBm) to 22 dBm. Per FCC rule, for the gain over 6 dBi ($30 - 6 = 24$ dB), we must reduce the transmit power by 1 dB for every 3-dB increase. Hence $24/3 = 8$ dB must be reduced from 30 dBm to 20 dBm. Refer to the link budget parameters in Table 14.3.

As seen from the calculation in Table 14.3, we have a 12-dB margin over and above the losses, which proves that not only did the link close, but also that a margin is left over. For this test, the equipment used was Linksys WRT54G. It can be observed that if the FCC PTP link rule were applied, then a higher transmit power or higher antenna gain can be used. Provided the equipment can

Table 14.3
Venezuela Linksys WRT54G Furthest Range to Date

Parameters	**Values for Outdoor Downlink**
Transmit power—P_{tx} (dBm)	22
Tx cable loss—L_{txc}(dB)	3
Tx antenna gain—G_{txant}(dBi)	30
Range desired—*d* (km)	270
Frequency—*f* (MHz)	2,400
Free Space Path Loss—*Lfspl*, with respect to Range (dB)	$= 32.45 + 20\log f + 20\log d$ $= 32.45 + 20\log 2{,}400 + 20\log 270 = 149$
Rx antenna gain—G_{rxant}(dBi)	30
Rx cable loss (dB)—L_{rxc}	3
Rx sensitivity—RX_{sens}(dBm)	= –85 dBm
Fade Margin (dB)	$= P_{tx} - L_{txc} + G_{txant} - L_{fspl} - L_{rxc} + G_{rxant} - RX_{sens}$ $= 22 - 3 + 30 - 149 - 3 + 30 - (-85) = 12$

support it, it can be argued even longer ranges can be realized. Clearly in this case, it appears that a direct line of sight targets the receiver. On the other hand, the margin just calculated is more than sufficient. I think the limitation in this case may very well be the hardware, more particularly, the Linksys (AP) transmit power being capped at 100 mW, and the practicality of using any higher-gain antenna than 30 dBi.

14.6 Coverage

An RF survey should always precede deployment. The RF survey shows where the signals are strong and where they attenuate, indicative of where a new AP is needed. The survey should be a simple one comprised of an AP transmitter and a portable RF receiver and recording device, preferably with one that ties the signal with the location as it records the data. A tool called WiSpy is recommended by [1] for surveying purposes, along with other measurement tools.

WiFi coverage, in order to be effective, must consider covering all areas, including where the service is needed, such as stairwells, break rooms, and building entrances, especially if the application includes phone conversation (VoIP). The link budget should also include body losses that can be significant for handset users. Coverage must have enough overlap for handoffs. The design also should take into account the fact that the radio-signal requirement for a phone application and a data application are as different as the locations in which they are used. Due mainly to body losses, and so on, the requirements are greater for voice applications [1].

To start your coverage deployment:

1. Separate each interfacing AP on the same floor by about 300 ft (90m).
2. Assign frequencies of channels 1, 6, and 9 corresponding to numbers 1, 2, and 3, respectively, for 802.11b, as shown in Figure 14.3. Residential locations use only one of the three channels (1, 6, or 9). It is a good idea to use one that is least used in the area.
3. For multiple floor applications, assume spherical coverage and take into account the floor losses, as well as the partition and solid wall losses. Refer to Figure 14.4.
4. Continue to use nonoverlapping channels, as shown in Figure 14.3.
5. Keep channel reuse distances as far apart as possible to improve capacity.
6. To estimate signal losses across floor and wall structures, make use of the RF survey results.

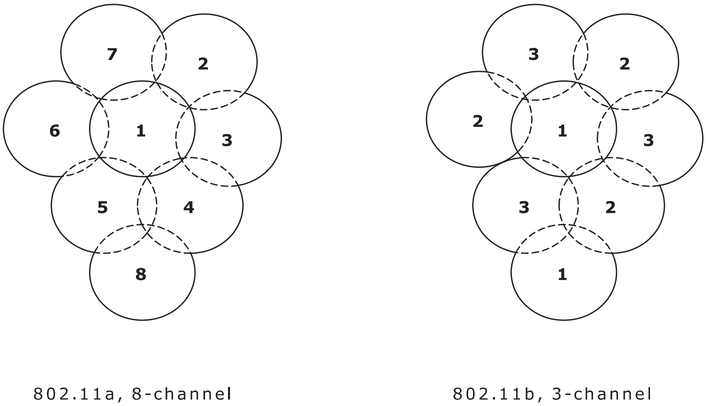


Figure 14.3 Eight-cell frequency assignment comparison.

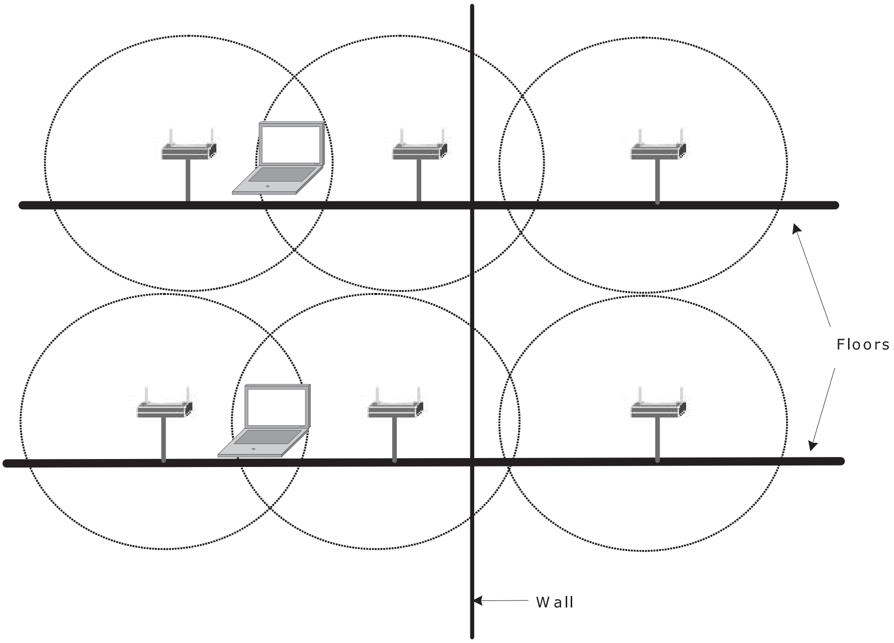


Figure 14.4 Interwall, interfloor coverage.

7. If the interwall signal is significantly attenuated, then you can use the overlapping channels—1, 2, 3, 4, and so on.
8. Special APs, depending on the interference activity, have dynamic channel reassigning features that must be exploited.
9. Simple interference-mitigating techniques to consider:

a. Add more APs to create a dominant signal.

b. Turn down the transmit power of APs, especially ones that are out further.

14.7 Capacity

Additional capacity in many cases requires additional APs, once the requirements for coverage are satisfied. The application type also derives the capacity need. While a data application cannot be catastrophically affected by insufficient capacity, the only symptom being slower Web browser response, phone conversations cannot tolerate irregular and, at times, excessive delays, without disruption.

A quick and back-of-the-envelope calculation of capacity is as follows. Using 802.11b, given 11 Mbps supplied by the AP, and given that a typical data rate used for a voice coder used is 64 kbps (unless it is compressed) for a voice application, then 64/11,000 = 5.8%. And taking into account any other overhead, we need a total of about 5% of the AP's 11 Mbps. Including any additional overhead for access and retries reduces the AP capacity to a maximum of 80% of the 11 Mbps, then 0.8/0.05 = about 16 simultaneous calls can be supported. Now, the number of calls that can be supported in the trunked or Erlang sense is higher, since not all calls will be up at the same time. For a data application for access to the Internet, browsing, e-mail, and so on, we require less bandwidth per call, or no more than 40 kbps. As a result, the number of simultaneous users will be higher.

See Table 14.4 for a comparison of capacity when downloading a 1-MB file.

In the following discussion, we rely largely on experimental outcomes [2] found in coverage and capacity experiments of 802.11a and 802.11b deployments.

Table 14.4
A Typical Download Speed for a 1-MB File, Using Various Modems

Modem Type	Data Rate	Duration to Download
Cable modem	1 Mbps	7 seconds
DSL modem line	256 kbps	32 seconds
Dialup modem	28.8 kbps	277 seconds

802.11g is at a higher data rate (54-Mbps peak) than 802.11b (11-Mbps peak). It is backward compatible to the latter and uses a different band and different modulation scheme.

802.11a delivers the same data rate as 802.11g, but occupies the 5-GHz spectrum.

The wider spectrum availability in the 5-GHz range results in more nonoverlapping channels, and makes 802.11a more compelling in the lower and upper two noncontiguous UNII bands (5.15–5.35 GHz and 5.725–5.825 GHz).

802.11a has a 300-MHz bandwidth total, as opposed to just over 80 MHz for 802.11b/g.

The number of nonoverlapping channels translates to 8–20 for 802.11a, as compared to just 3 for 802.11b. There is an interplay between increasing the number of sites (APs) and the number of nonoverlapping channels. Cochannel interference and throughputs are also dependent on these interplays. See Figure 14.5, which shows the reuse of channels and interference potentials. For the same number of sites used in 802.11a and 802.11b, the cochannel interference is less in the case of the former than of the latter.

> Summary of the outcomes, among others, established in the experiments [2] conducted:
>
> - 802.11a has a similar range as 802.11b in a typical office environment up to 225 ft. However, the data-link rate is 2–5 times more in favor of 802.11a.

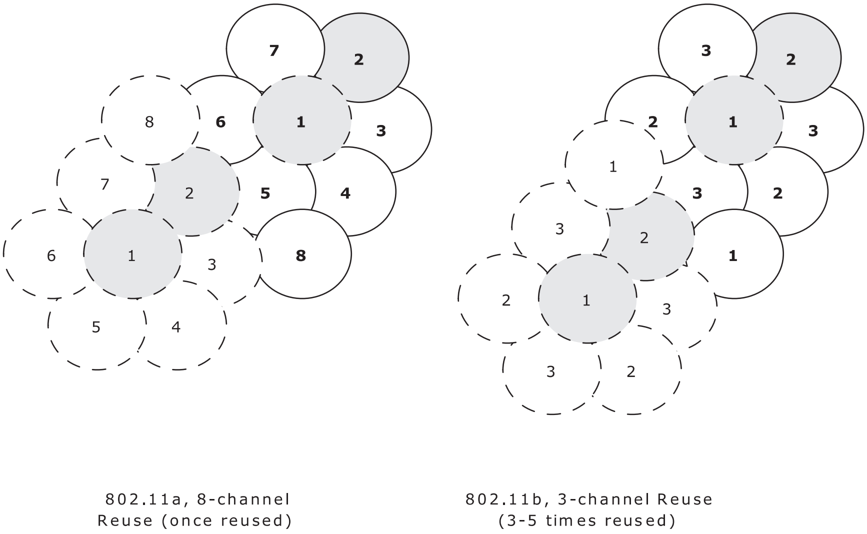


Figure 14.5 Sixteen-cell reuse comparison.

- 802.11a throughputs are at least a factor of 2–4.5 times larger than 802.11b systems in a typical office environment of up to 225 ft.
- For an average cell radius of 65 ft, an 8-cell system of 802.11a has 8 times the average cell throughput, compared to 802.11b system.

802.11b shows five cochannel interferences for the three channels. That is, in the first 8-site deployment (refer to Figure 14.5) except imagine eight-cell deployment in each case, as opposed to 16, as shown in the figure):

Channel 1 has one interferer.

- Channel 2 has two interferers.
- Channel 3 has two interferers. That is a total of five interferers.

On the other hand, 802.11a experiences no source of interferers in the eight-site system. As a result, 802.11a has eight times the throughput advantage, as [2] established. 802.11a has eight times the system capacity of 802.11b for an eight-cell WiFi deployment.

Further, when the 16-site coverage is considered (see Figure 14.5), in the case of 802.11a, the channels are reused only once, whereas the three-channel frequencies in 802.11b are used three to five times for the same coverage area. Interferences are that much more likely when channels are reused numerous times. As the area of coverage increased, the three channels in the 802.11b must be used even more than five times. It should be pointed out this comparison is based on the fact that 802.11a uses only eight channels. The good news is that up to 20 nonoverlapping channels can be used and are available in 802.11a. Granted, some equipment does not necessarily take advantage of the availability of this wider spectrum, or for various other reasons, it does not get implemented.

Not surprisingly, [2] also established in their experiment that the overall system capacity is higher for fewer numbers of APs (sites), with the 802.11a coverage build-out than with that for 802.11b [1]. In addition, according to the results, the system capacity difference is higher for 802.11a than for 802.11b by up to 100 Mbps for the same area covered.

Another somewhat counterintuitive, but correct conclusion drawn from experiments in [2] is that 802.11a and 802.11b have similar ranges of up to 225 ft in an office environment, and at the same time, 802.11a has a 2–5 times higher data rate and throughput. On account of its higher frequency, one would think 5 GHz would have more loss than 2.4 GHz by at least a factor of 2. However, in a confined environment, higher frequencies perform better than the lower frequencies, and so they somewhat mitigating the loss.

14.8 Security Gaps and Solutions

An illegitimate BS steals the ID of a legitimate BS in order to then communicate with a MS to extract private information or join in a conversation. Suppressing SSID and filtering MAC, considered a security solution in WiFi, has proved to be ineffective for WiMAX. It is ineffective because initially, when the client requests an SSID, it is broadcast over the air in response to that request. And a MAC address can be forged, too.

The widely known and, for the most part, discredited wired equivalent privacy (WEP) created a fallback response of a new standard called IEEE 802.1x to help authenticate and secure both wireless and wired LANS. The wildcard with 802.1x protocol is interoperability.

WEP encryption can protect against casual snooping, but it may also produce a false sense of security, as freely available tools, such as AirSnort, recover WEP encryption keys. The software analyzes in short order several millions of encrypted packets enough to identify which part is a password. There are other more advanced and efficient tools, with a higher success rate, that can do the trick with fewer packet samples.

The newer WiFi protected access (WPA) and IEEE 802.11i (WPA2) encryption standards do not have any of the serious weaknesses of WEP encryption.

14.8.1 IEEE 802.1x

IEEE 802.1x is an IEEE standard for port-based network access control. It is part of the IEEE 802.1 group of networking protocols. It provides an authentication mechanism to devices wishing to attach to a LAN port, either establishing a point-to-point connection or preventing access from that port if authentication fails. It is used for most wireless 802.11 access points and is based on the extensible authentication protocol (EAP).

Mostly used in such operating systems as Windows XP, 802.1x is a standard for encapsulating extensible authentication protocol (EAP) over a wired or wireless local area network (WLAN).

EAP, among other things, provides the framework and format for authentication.

Table 14.5
Basic 802.1x Format

802.1x (Header)	EAP (Payload)

In many ways, 802.1x is similar to point-to-point protocol (PPP). In a situation where the extra and unneeded overhead and complexity of PPP is not needed, 802.1x is a better choice to pass EAP messages.

To understand the context in which 802.1x functions. please refer to Figure 14.6.

The wireless client that needs to be authenticated and asks for service is called a *supplicant*. The server, typically a RADIUS server, that does the authentication is called, not surprisingly, the *authentication server*. The device that sits between the supplicant and the authentication server, similar to AP, is called the *authenticator*. One compelling advantage of 802.1x is that it functions on the AP and does not require that much intelligence be resident on the same. The intelligence can be elsewhere, either in the wireless client or on the server. For this reason, 802.1X is particularly well suited for wireless LAN applications,

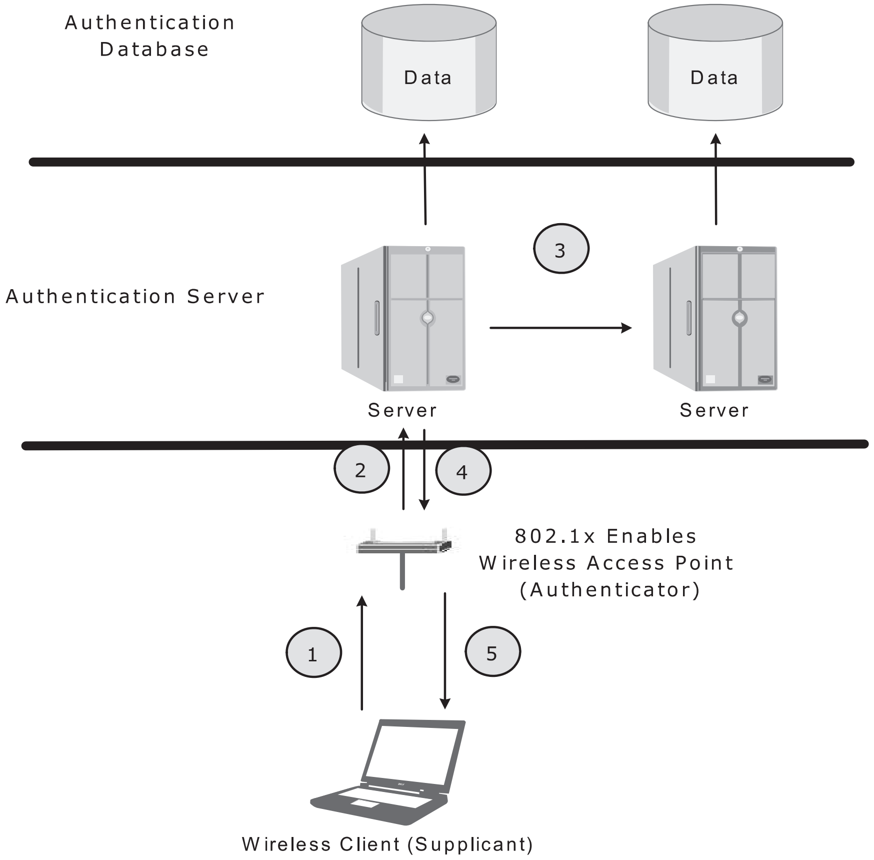


Figure 14.6 802.1x authentication.

because it requires very little processing power on the part of the authenticator or wireless AP.

WiFi access-point vendors now use 802.11i, which implements 802.1x for wireless access points to address the security vulnerabilities found in WEP. The authenticator role is either performed by the access point itself via a preshared key (referred to as WPA2-PSK) or, for larger enterprises, by a third-party entity, such as a RADIUS server. This provides for client-only authentication or, more appropriately, strong mutual authentication using protocols such as EAP-TLS.

Also, 802.11i RSN (robust security network) protocol uses 802.1x to authenticate wireless devices to the network and to provide the dynamic keys it requires.

14.8.2 802.1x Authentication Steps

Prior to being granted access, the port of the switch the supplicant interfaces with on the authenticator is set to the unauthorized state. In this state, 802.1x traffic is allowed [3]. Other traffic, such as DHCP and HTTP, are blocked at the data-link layer. Refer to Figure 14.6.

The following describes the 802.1x authentication steps:

1. Supplicant sends request to be authenticated to the wireless AP (or 802.1x-enabled switch). 802.1x Supplicant provides client capability through computers, routers, switches, PDAs, or IP phones.
2. Wireless AP retrieves the EAP payload from the 802.1x message, repackages the authentication request and sends it to authentication server via RADIUS protocol. The RADIUS server must be compatible with EAP and the 802.1x standards. The authenticator (switch or router) is the middleman for relaying EAP received in 802.1x packets to an authentication server by using RADIUS to carry the EAP information.
3. The server may proxy the request or refer to the authentication database.
4. If authentication is confirmed, the server informs AP.
5. AP then informs the supplicant of the access grant. (At this stage, AP is allowed access to resources located on the protected side of the network. The port interface at the authenticator is set to the *authorized* state and normal traffic is allowed.)

14.8.3 The Significance of EAP

Each protocol that uses EAP defines a way to encapsulate EAP messages within that protocol's messages. In the case of 802.1X, this encapsulation is called EAP over LANs (EAPOL).

Extensible Authentication Protocol (EAP) is a universal authentication framework mainly used in the PTP connections of a wireless network. It can be used with both wired and wireless LANs, and additionally, it has been adopted with WPA and WPA2 standards. EAP, while not providing a specific authentication mechanism, however, provides an authentication framework, common functions, and a negotiation of the desired authentication mechanism called *methods.* There are currently dozens of EAP methods available. Some common methods are EAP-TLS, EAP-IKEv2, EAP-SIM, and EAP-AKA.

When EAP is invoked by an 802.1X-enabled network-access-server (NAS) device, such as an 802.11 a/b/g wireless access point, modern EAP methods can provide a secure authentication mechanism and negotiate a secure pairwise master key (PMK) between the client and NAS. The PMK can then be used for the wireless encryption session, which uses Temporal Key Integrity Protocol (TKIP) or Countermode/CBC-MAC Protocol (CCMP), based on AES encryption.

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[1] Geier, J., "How to Assign 208.11b/g Access Point Channels," Wireless-Nets, Ltd., 2008, http://www.wireless-nets.com.

[2] Chen, J., and J. Gilbert, *Measured Performance of 5-GHz 802.11a Wireless LAN Systems*, Sunnyvale, CA: Atheros Communications, 2001.

[3] Roshan, P., "802.1x Authenticates 802.11 Wireless," *Network World,* 2001, http://www.networkworld.com.

15

UMTS and WiMAX Comparison

15.1 Overview

Universal mobile telecommunications service (UMTS) is a third generation (3G) emerging technology along the path of global system for mobile's (GSM's) telecommunication evolution. The evolution to UMTS took several steps. The 3G UMTS traversed an evolutionary path from its antecedent technologies, starting with the globally successful digital mobile telecommunication system, GSM, to general packet radio service (GPRS), then to enhanced data rates for GSM evolution (EDGE), to finally UMTS.

GSM is a globally successful digital mobile telecommunication system, with more users in the world than any other technology. GPRS, considered 2.5 generation (2.5G), is the packetized application arm of GSM. GPRS is assumed to deliver a peak data rate of about 110 kbps.

EDGE, technically classified as 2.75G, delivers a higher data rate than GPRS, from 230 kbps to 470 kbps, depending on the number of time slots used. In order to achieve this higher data rate, EDGE incorporated the 8PSK modulation technique.

Each of the evolutionary steps toward UMTS (GSM > GPRS > EDGE > UMTS) required it to support its legacy network [1]. Also, at each step, the technology incorporated enhancing mechanisms, such as subsystem modifications and software changes, to meet demands and stay competitive with other technologies. These enhancements were typically for the purpose of improving data communication, efficient data rates, and service flexibility. For example, in so far as GSM is concerned, enhancements included value-added services (VAS), intelligent network (IN), and high-speed circuit-switched data (HSCSD). In turn, HSCSD had its own improved version with enhanced circuit switch data

(ECSD). GPRS had its own enhancements as well, for example, enhanced GPRS (EGPRS).

Then the GSM/EDGE evolution path, by the 3GPP release 5, allowed a stand-alone GSM/EDGE radio access network (GERAN) to the core network (CN), which interfaced to the CN and provide the same services as the full-blown UMTS terrestrial radio access network (UTRAN) in UMTS. In this configuration, the GERAN and UTRAN are supposed to be connected independently, and each provided the same services (interactive, streaming audio, and video). Clearly, in order to connect to a 3G, CN GERAN had to go through a great transformation to be able to function as UTRAN. This effectively allowed two independent all IP-based network paths to be smoother. Figures 15.1 through 15.5 show the evolutionary path of UMTS from the network standpoint. Hence, during transitions in a general sense, a dual-mode phone capable of accessing GSM and GERAN is employed, whereas in 3G, yet another dual-mode phone is used for accessing both GSM and UTRAN.

UMTS is IP based and better suited for data communication and compatibility with the Internet. Those similarities to WiMAX and exact reasons are why UMTS is being discussed here. Moreover, not only are WiMAX and UMTS similar from an architecture standpoint, but they also can interface with each other to work together. Of course, either technology interfaces with the Internet.

The standardization areas of UMTS based on the third generation partnership project (3GPP) technical specification include the following sections:

- Radio access network;
- Core network;
- Terminal;
- Services and systems aspects;
- GERAN.

In this book, our main theme is physical air interface and RF design. Following our discussion of the fundamentals of UMTS technology, especially as it relates to the WCDMA radio interface frame structures, link budgets, and capacity computations will be explored in Sections 15.8 and 15.9. The other impetus of bringing UMTS into this discussion is to be able to appreciate the level of complexity involved when a non-IP system is made to evolve toward a full-IP system. Given the stark similarity between WiMAX and UMTS, the former is not going through the complex steps of adapting to an IP-based system; rather it was created as an IP-based system from the beginning.

The workaround or extensions—software or hardware or both—can be daunting at times, and the finished product may not be as efficient as it was intended to be.

At the present time, when nearly all technologies are marching toward having the characteristics of a high data rate, simple, IP-based, efficient, and low-maintenance technology, the comparison gives appreciation for writing specifications and implementations from scratch, rather than forcing a change on an existing system as demands and requirements dictate.

At the outset, it should be pointed out UMTS is in many ways a marriage between the traditional GSM higher layers and the wideband CDMA (WCDMA) physical air interface layer. The physical layer of UMTS, wideband code division multiple access (WCDMA), uses direct sequence spread spectrum (DSSS).

In the 1990s, when CDMA technology burst onto the scene, it was obvious that it brought with it several appealing features that qualified it to be a fresh entrant with a convincing technology. CDMA also inspired the incorporation of its compelling advantages into UMTS. Some of the highlights of CDMA worth mentioning are:

- The implementation of code division implies that because the transmitted signal is made to appear as noise, resulting from a pseudonoise code application, (hence, the name code division) signals can be detected well below the noise floor just by applying the right code. In the non-CDMA design, the receiver's sensitivity levels must be low enough to guarantee the detection of small signals close to the noise floor. (Generally receiver sensitivity is given by Receiver Sensitivity = Total Noise + Noise Figure + SNR. As shown later, a receiver sensitivity shown in the link budget in Table 15.2 is about 119 dBm, way above the noise floor.)
- As a byproduct or as the result of the above, CDMA does not require a lot of transmit power, a benefit for interference control.
- Capacity advantage.
- Voice-quality advantage.
- While using the handset, lower electromagnetic exposure to the human head and body.
- Multipath resolutions using a rake receiver.
- Soft/softer handoff gains.

Now, incorporating these physical-layer benefits into GSM puts UMTS at a tremendous advantage. Indeed, it can be said UMTS is a marriage between CDMA and GSM in many ways.

It should be mentioned that fundamental CDMA technology that helped inspire the UMTS design has had to evolve itself to CDMA2000, in order to meet the ever increasing demand for an efficient data rate and IP-based technology.

15.2 GSM-to-UMTS Evolution

In Figures 15.1–15.5, you will observe a steady change in the network configuration, starting with the traditional GSM. In each advancing technology, over the previous one, there was a requirement to support the legacy network users—a key source of complication in an evolving system. GPRS was the next implementation in which both hardware and software changes were implemented.

However, the GPRS stage of GSM's evolution was a milestone, because that was where the packet-based architecture was laid out (Figure 15.2). It can be seen from the figures that GPRS architecture and its components carry over, way into the final UMTS 3G implementation. The next stage was EDGE implementation. The main impetus for this change was an improved data rate by improving mainly the modulation scheme. Software updates pertaining to EDGE were also required (Figure 15.3).

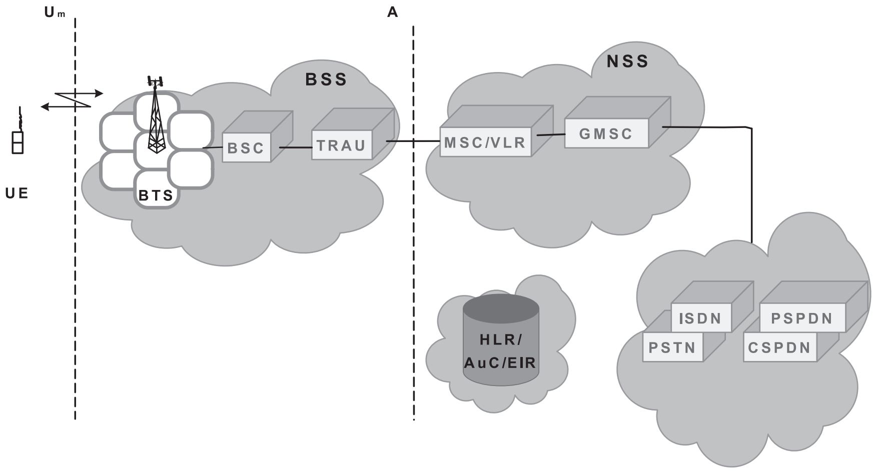


Figure 15.1 Basic GSM architecture.

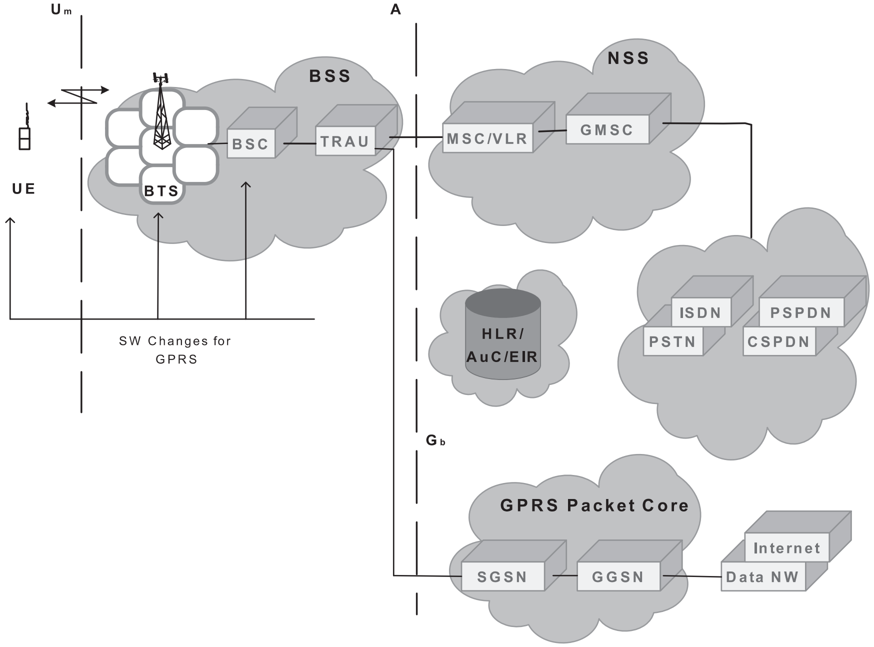


Figure 15.2 GPRS architecture.

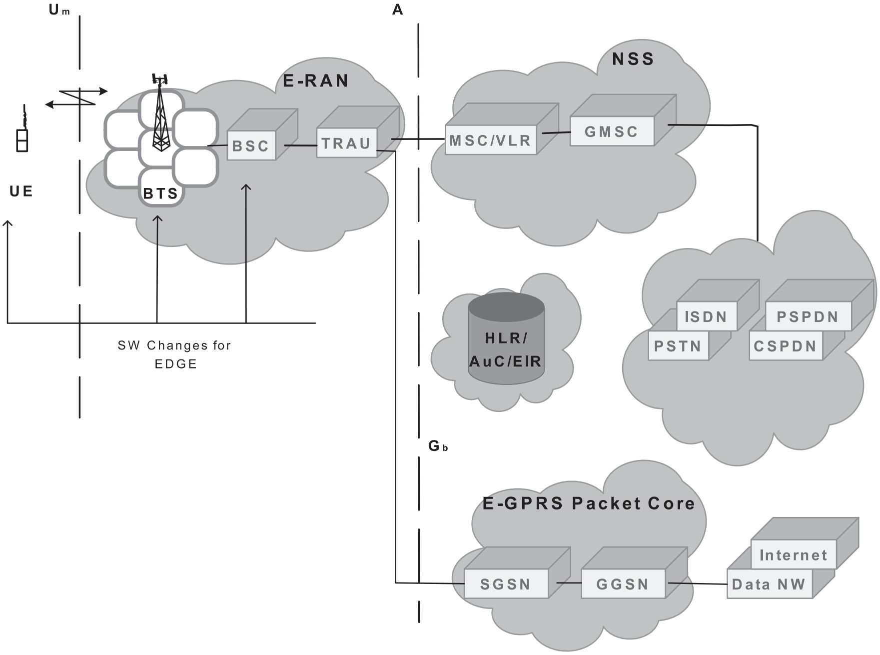


Figure 15.3 EDGE architecture.

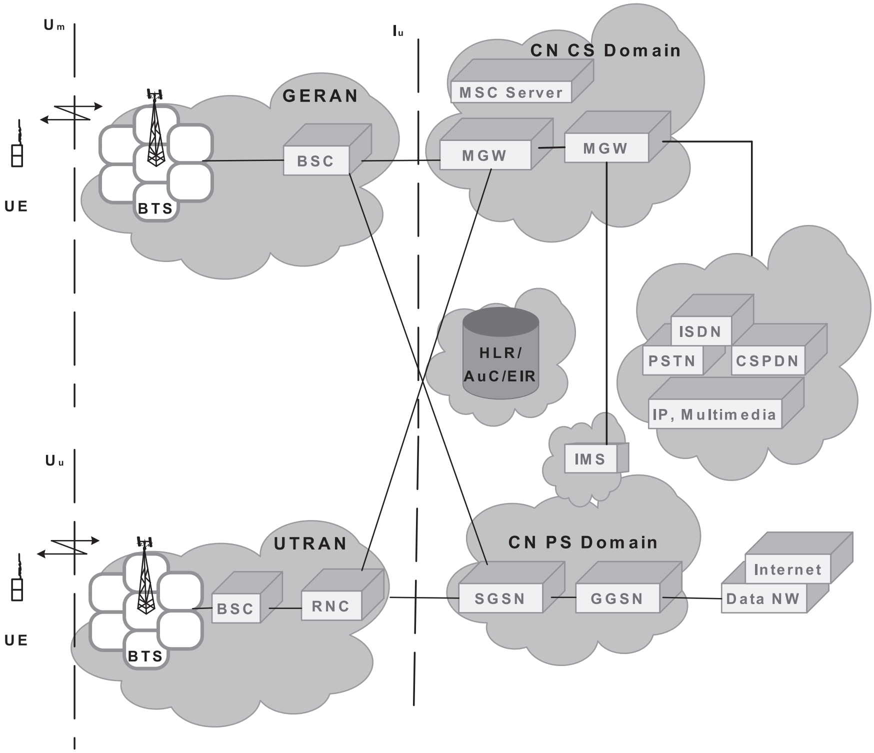


Figure 15.4 GERAN architecture.

The next stage in the path of GSM evolution before UMTS was GERAN implementation. GERAN was another drastic implementation created by splitting the radio access network into two: one interfacing with 2G GSM CN and the other with 3G UMTS CN. This split maximizes the commonality between UTRAN and GERAN, while maintaining backward compatibility with GERAN. The key is that now both UTRAN and GERAN have a common interface to the 3G CN.

Such actions in development, though significant and complex in implementation, allowed a smoother transition to UMTS. The last improvement over UMTS was high-speed downlink packet access (HSDPA) implementation. In HSDPA, the UMTS architecture and interfaces between components remained for the most part unchanged. The significant change in HSDPA was the air interface. That is, the interface between the UE and the node B/radio network controller (RNC). The capacity gain in HSDPA implementation, as high as 14 Mbps, is explained in Section 15.9.

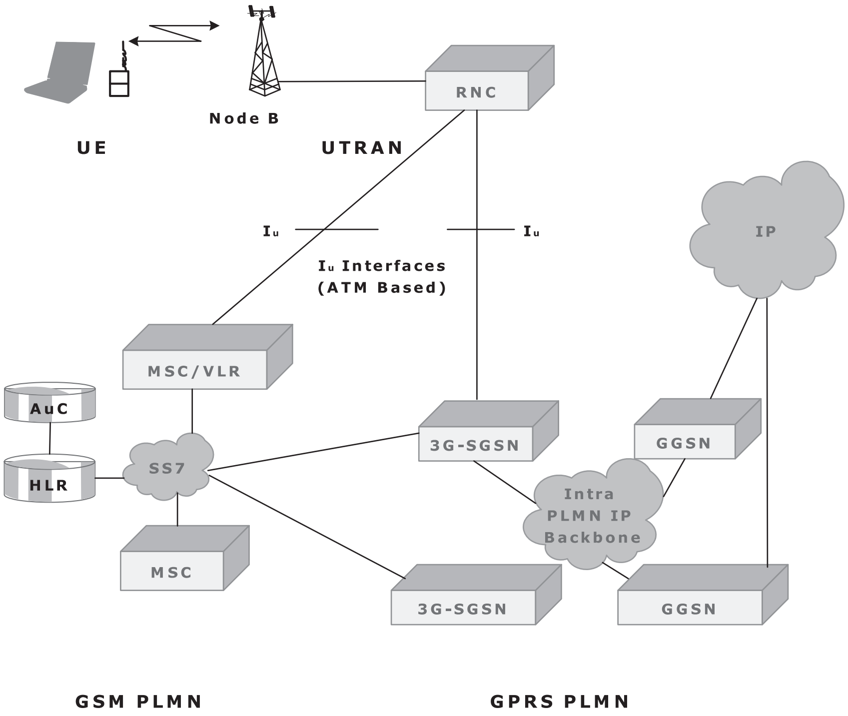


Figure 15.5 3G UMTS networks.

15.2.1 Key Component Definitions

Mobile Switching Center/Visitor Location Register

The mobile switching center (MSC) cross-connects calls, and supports pulse-code mode modulation (PCM) and interworking function (IWF) capability for fax and data calls.

MSC does all the switching coordination for all the mobiles in its area. In addition, it performs intermediate connections for the visitor location register (VLR), handling any interactions with mobiles that it needs.

Home-Location Register

Home-location register (HLR) is the central database in the UMTS network. When a subscriber attaches to VLR, it then updates HLR on the current location of the subscriber.

Authentication Center

Authentication center (AuC) is another database that works closely with HLR. In the AuC database subscriber security authentication information is maintained. The security key generated in the AuC must match that which is in the subscriber identity module (SIM).

Third Generation–Serving GPRS Support Node

Third generation–serving GPRS support node (3G-SGSN), like VLR, tracks the UE and validate/authenticate in the same way. ATM is used to connect 3G-SGSN to UTRAN (Figure 15.5).

Gateway GPRS support node (GGSN) serves as a gateway between UMTS PLMN and the IP network to support the UMTS service. UMTS routes data between the packet data network to the UE and vice versa. Once UE activates its packet address, it is registered with GGSN. In the IP backbone, 3G-GGSN and GGSN interconnect. GGSN also interfaces with HLR to get UE address/location.

UMTS utilizes strong ATM backbones that most service providers use, and, at the same time, provide service grade differentiation via QoS on the ATM (transport protocol). Since IP was designed for best-effort delivery service, it does not distinguish between different users. While this works okay for nonreal-time applications, for real-time applications, such as streaming audio and video service, differentiations must be incorporated.

15.3 UMTS Architecture’s Similarities with WiMAX

Please refer to Figure 15.6, which shows the stark similarity between the WiMAX network and the UMTS network. Clearly, in either case, the UE or the SS gets connected to the IP via two entities in the middle: generically speaking, the radio access network and the core network [2].

The first entity that the UE/SS encounters over the air interface is the radio access network (UTRAN for UMTS and ASN for WiMAX). The second entity is the core network (CN in case of UMTS and CSN for the case of WiMAX).

It is also easy to see that the two networks can communicate with each other through the IP network.

15.4 Physical Layer Specifications

Table 15.1 organizes the UMTS WCDMA physical layer specification and parameters. For comparison purposes and to highlight the differences, a couple of IS-95- related parameters are shown—bandwidth and multipath resolution.

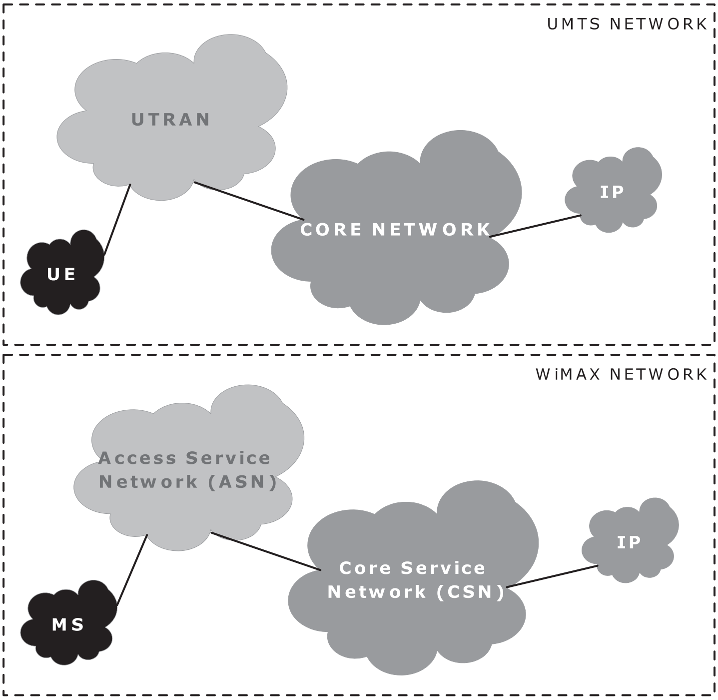


Figure 15.6 WiMAX and UMTS comparison.

Table 15.1
UMTS WCDMA Physical Layer

Channel Bandwidth (MHz)	5	**Super Frame Duration (ms)**	720
Chip Rate (Mcps)	3.84	**Frame Duration (ms)**	10
Multipath Resolution (μs)	0.2	**Slot Duration (ms)**	0.667
IS-95 Channel Bandwidth (MHz)	1.25	**Frames in Super Frame**	72
IS-95 Multipath Resolution (μs)	0.8	**Slots in A Frame**	15
Maximum Data Rate Supported (Mbps)	2	**Separation of Conversation in Both Directions**	Channelization code
Power Control Rate (Hz)	1,500	**Increasing Data Rates**	Spreading code (orthogonal variable spreading factor)
Support FDD Mode?	Yes	**Number of Spreading Factors**	4–512
Supports TDD Mode?	Yes	**Number of Scrambling Codes**	16.8·106

15.5 WCDMA Physical Layer Radio Frame

UMTS/WCDMA utilizes two types of access methods: one in code-division multiple access of the air interface, in which users can access the same system with different codes, and the other is the UMTS frame-structure time-division access technique of the slot.

The WCDMA transmission is structured into frames [3], slots, and channels in order to deliver data in an effective fashion. Please refer to Figure 15.8, which shows the WCDMA physical layer radio layer. As shown, a super frame with 720-ms duration comprises 72 frames. The duration of each is 10 ms (72 frames·10 ms = 720 ms). A single frame has 15 slots, each with a duration of 0.667 ms. Each time slot contains 2,560 chips, and thus, the entire 10-ms frame contains 38,400 chips (2,560·15 ms).

There are three groups of channels:

- Logical channels;
- Transport channels;
- Physical channels.

Logical channels in UMTS are further categorized into control and user data channels, and they define the way in which the data will be transferred. The

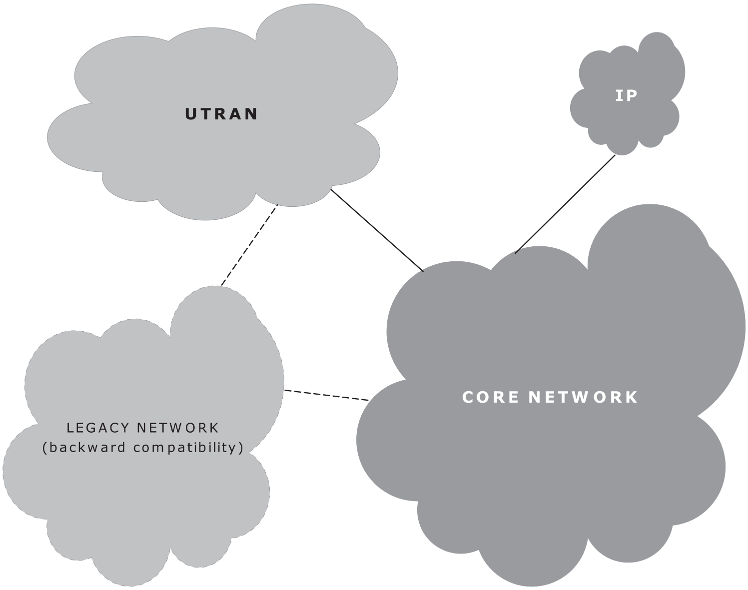


Figure 15.7 UMTS network clouds.

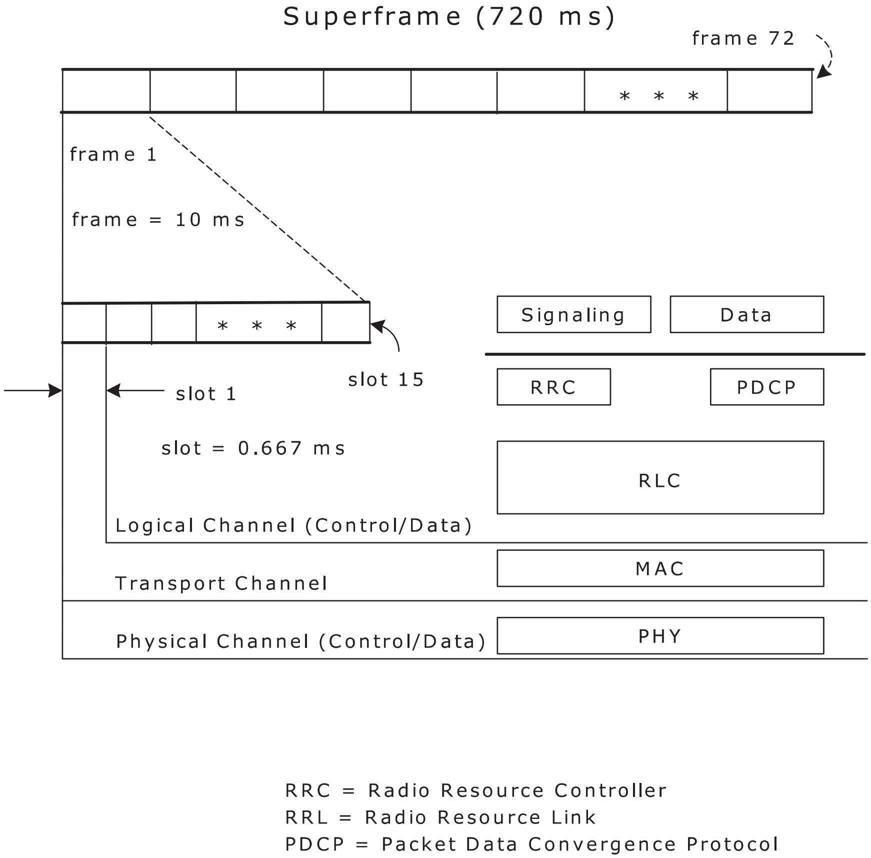


Figure 15.8 WCDMA physical layer radio frame.

transport channel also defines the way in which data is carried or transferred. The physical channel carries the payload and also specifies the signal characteristics. The physical channel also determines who gets the information.

There is also a mapping between the transport and physical channels [4]. Figure 15.9 depicts the transport channel to physical channel mapping.

15.6 Channel Definitions and Functions

15.6.1 Uplink and Downlink Channels in WCDMA

Channels are also classified according as downlink, uplink, bidirectional, dedicated, common, control, or traffic channels. Also, some are referred to as indicator, random, broadcast, shared, paging, uplink, downlink, primary, secondary, synchronization, and acquisition channels, to emphasize their purposes.

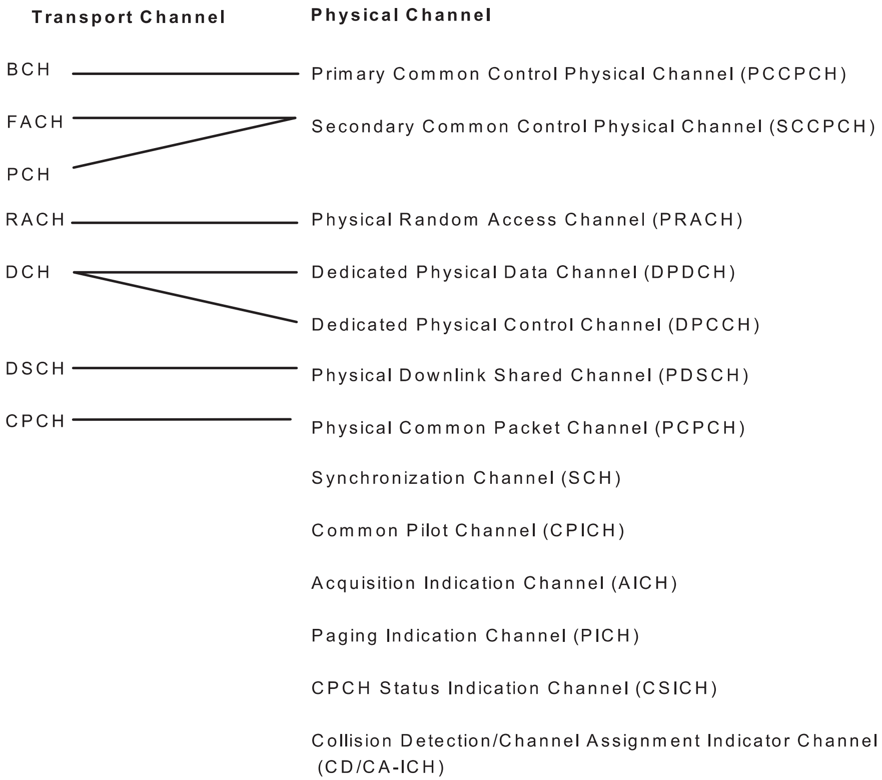


Figure 15.9 Transport channel-to-physical channel mapping.

However, the major classification remains common, dedicated, traffic, or control, all subsumed in one form or another into physical, transport, and logical channels.

Refer to the following definitions of the channels.

Logical Channels:

Broadcast Control Channel (BCCH) (downlink)—broadcasts information to UE relevant to the cell, such as radio channels of neighboring cells, and so on.

Paging Control Channel (PCCH) (downlink)—associated with the PICH and used for paging messages and notification information.

Dedicated Control Channel (DCCH) (uplink and downlink)—used to carry dedicated control information in both directions.

Common Control Channel (CCCH) (uplink and downlink)—a bidirectional channel used to transfer control information.

Shared Channel Control Channel (SHCCH) (bidirectional)—a bidirectional channel found only in the TDD form of WCDMA/UMTS, where it is used to transport shared-channel control information.

Dedicated Traffic Channel (DTCH) (uplink and downlink)—a bidirectional channel used to carry user data or traffic.

Common Traffic Channel (CTCH) (downlink)—a unidirectional channel used to transfer dedicated user information to a group of UEs.

Transport Channels:

Dedicated Transport Channel (DCH) (uplink and downlink)—used to transfer data to a particular UE. Each UE has its own DCH in each direction.

Broadcast Channel (BCH) (downlink)—broadcasts information to the UEs in the cell to enable them to identify the network and the cell.

Forward Access Channel (FACH) (downlink)—carries data or information to the UEs that are registered on the system. There can be more than one FACH per cell, as they may carry packet data.

Paging Channel (PCH) (downlink)—carries messages that alert the UE to incoming calls, SMS messages, data sessions, or required maintenance, such as reregistration.

Random Access Channel (RACH) (uplink)—carries requests for service from UEs trying to access the system

Uplink Common Packet Channel (CPCH) (uplink)—provides additional capability beyond that of the RACH and fast power control.

Downlink Shared Channel (DSCH) (downlink)—can be shared by several users and is used for data that is bursty in nature, such as that obtained from Web browsing.

Physical Channels:

Primary Common Control Physical Channel (PCCPCH) (downlink)—continuously broadcasts system identification and access control information.

Secondary Common Control Physical Channel (SCCPCH) (downlink)—carries the forward access channel (FACH) providing control information, and the paging channel (PACH) with messages for UEs that are registered on the network.

Physical Random Access Channel (PRACH) (uplink)—enables the UE to transmit random access bursts in an attempt to access a network.

Dedicated Physical Data Channel (DPDCH) (uplink and downlink)—used to transfer user data.

Dedicated Physical Control Channel (DPCCH) (uplink and downlink)—carries control information to and from the UE. In both directions, the channel carries pilot bits and the transport format combination identifier (TFCI). The downlink channel also includes the transmit power control and feedback information (FBI) bits.

Physical Downlink Shared Channel (PDSCH) (downlink)—shares control information with UEs within the coverage area of node B.

Physical Common Packet Channel (PCPCH)—specifically intended to carry packet data. In operation, the UE monitors the system to check if it is busy. If not, it then transmits a brief access burst. This is retransmitted, if no acknowledgement is gained, with a slight increase in power each time. Once node B acknowledges the request, the data is transmitted on the channel.

Synchronization Channel (SCH)—used in allowing UEs to synchronize with the network.

Common Pilot Channel (CPICH)—transmitted by every node B so that the UEs are able to estimate the timing for signal demodulation. Additionally, they can be used as beacons for the UE to determine the best cell with which to communicate. For example, this channel can be used in the determination of a specific scrambling code within a group of eight. By performing a correlation between this channel and each of the scrambling codes in the group, the specific code is identified.

Acquisition Indicator Channel (AICH)—used to inform a UE about the data channel (DCH) it can use to communicate with node B. This channel assignment occurs as a result of a successful random access service request from the UE.

Paging Indication Channel (PICH)—provides information to the UE to be able to operate its sleep mode to conserve its battery when listening on the paging channel (PCH). As the UE needs to know when to monitor the PCH, data is provided on the PICH to assign a UE a paging repetition ratio to enable it to determine how often it needs to wake up and listen to the PCH.

CPCH Status Indication Channel (CSICH)—appearing only in the downlink, CSICH carries the status of the CPCH and may also be used to carry some intermittent or bursty data. It works in a similar fashion to PICH.

Collision Detection/Channel Assignment Indication Channel (CD/CA-ICH)—present in the downlink, CD/CA-ICH is used to indicate whether the channel assignment is active or inactive to the UE.

The common transport channels needed for basic network operation are RACH, FACH, and PCH, while the use of DSCH and CPCH is optional and can be decided by the network.

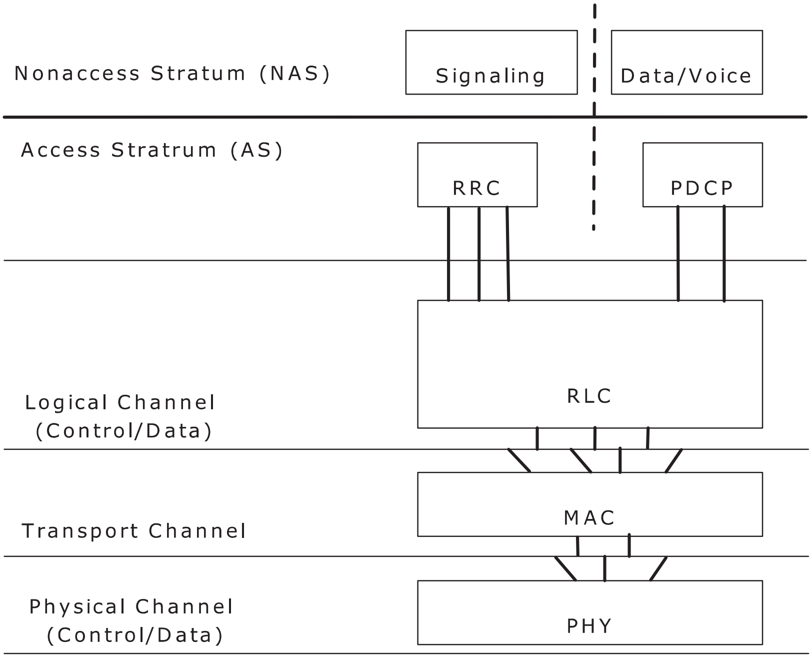
15.6.2 Channel Mappings

There is a mapping between logical and transport channels in both uplink and downlink directions. As shown in Figures 15.9 and 15.10, there is also a mapping between transport and physical channels [4, 5].

15.6.3 Essentials of UE-Node B Air Interface Communication

A typical UE interaction with the network is through the air interface, also called Uu interface, to node B in the UTRAN. The following are typical and major phases a typical UE encounters:

1. Initial power turn on to idle mode;
2. Call setup (requires the authentication of the UE radio bearer setup);
3. Handover (requires a power measurement of candidate cells);
4. Call release back to idle mode.



RRC = Radio Resource Controller
RRL = Radio Resource Link
PDCPP = Packet Data Convergence Protocol

Figure 15.10 Mapping of channels.

The following illustrates the point and demonstrate the functions of the various channels in the context of channels already discussed.

When powered on for service, the UE automatically receives system information in the air interface from node B (BTS) via the BCCH/BCH channel. BCH is a transport channel used to transmit information specific to UTRAN. This includes information such as random access code and access slots.

UE must decode the broadcast channel before registering. Following system parameter acquisition on the downlink, the next order of business for the UE is one of several actions, depending on whether the UE performs location update, registration, or call setup. The registration, location update (when UE changes one location area to the other), or call setup it needs to use the random access channel (RACH)—a transport channel—is mapped to the random access channel (PRACH)—(the physical channel). Refer to Figure 15.9 showing transport channel to physical channel mapping. PRACH is used to carry random access transmissions as the UE does when it initiates a call. The random access procedure is based on slotted ALOHA.

> A word or two about random access technique:
>
> Random access protocol is a protocol that implements the random transmission of packets from several nodes, each node having the same probability of transmission. The random access protocol assumes that the data rate is constant for all transmission of each node. However, when two or more packets are transmitted to the same destination at the same time, collisions occur, causing all packets to be lost. In order to solve this collision problem, MAC protocols allow for collision detections. The random access MAC protocols specify how to detect collisions and how to recover from these collisions. Recovery is usually through delayed retransmission of the packets.
>
> The slotted aloha is an example of the MAC protocol. The slotted aloha is a synchronized protocol, having slots of equal-sized intervals of time. Each node independently transmits with probability p. The throughput, if it were plotted against an offered load, looks similar to a slightly askew normal or Gaussian function.

The UE, using a nominal power, then transmits the PRACH preamble in order to get the attention of the cell. If the cell does not acknowledge it, the UE increases the transmit attempt's power again. This open-loop-power-control attempts to get the attention of the cell continue with more and more power than the last one in each attempt, until acknowledged. Once the cell acknowledges the receipt of preamble message, the UE then sends a PRACH message at the power level just acknowledged. Following this, the UE registers with the cell to be paged in case of incoming calls. It can then go on sleep mode (to conserve battery power, the receiver is shut down), from whence it periodically wakes up to monitor a page for an incoming call or place a call on the network.

Synchronization is a major component necessary before any meaningful one-to-one communication commences. This happens fairly early after the UE powers on, and there are various levels and needs for synchronization. In order to achieve timing synchronization, the UE first searches, even at the moment of startup, the strongest synchronization (SCH) channel it can find. The strongest signal received also determines the cell to which the UE will most likely lock. However, the UE must first achieve synchronization on numerous other levels. There are two SCH channels—primary and secondary. The primary SCH channel is a periodic signal to every slot (0.667 ms) and is good enough to derive a timing error. But in order to achieve chips, symbols, and slot synchronization, the secondary SCH, which is periodic every frame (10 ms), is needed. Having achieved chips, symbols, frame, and slot synchronization still doesn't qualify the UE to achieve a scrambling code that will allow it to descramble its data. A scrambling code is assigned by a cell to each UE, and it separates various users in the service area of the cell. Without a scrambling code, the UE cannot descramble its own data from the rest.

In order to achieve synchronization and identify the scrambling code, the UE needs two pieces of information:

1. The secondary SCH channel, which will tell the UE which scrambling-code group it belongs to, out of 512 possible scrambling-code groups. In each group there are eight scrambling codes.
2. In order to identify the one scrambling code out of the eight possible codes, the UE performs a correlation of each with the CPICH pilot channel. The highest correlation result identifies the right scrambling code. After this procedure, the UE is ready to demodulate and descramble data sent to it from the cell with which it is communicating.

15.7 Channelization and Spreading

15.7.1 Spreading and Scrambling Codes

Spreading codes (channelization codes) are orthogonal codes that are also called orthogonal variable spreading factor (OVSF) codes. OVSF codes are similar to Walsh codes in CDMA, and they separate channels in both directions, conversations being handled by the same base station or sector [3, 5].

Scrambling codes are long, random-like, 10-ms-lasting codes that separate transmit sectors or user equipment (UE) in the downlink and uplink, respectively.

Scrambling codes for the cell are assigned during the cell planning phase, while the spreading codes for the UE are assigned during setup by the radio network. Scrambling codes are termed pseudonoise (PN) codes in both CDMA and CDMA2000 technologies.

Figures 15.11–15.14 show scrambling and channelization (spreading) codes.

Spreading or OVSF Codes OVSF codes are used for the separation of different physical channels within a cell (downlink) or a mobile terminal (uplink).

Uplink OVSF There are only OVSF = 256 to differentiate physical channels from within a mobile terminal on the uplink. Similar to CDMA Walsh code, they channelize the various physical channels.

Downlink OVSF There are only OVSF = 512 to differentiate physical channels from within a cell or a sector on the downlink. Similar to CDMA Walsh code, they channelize the various physical channels.

Scrambling Codes And then there are codes to differentiate within cells and within MTs. They are called scrambling codes.

Downlink Scrambling Codes WCDMA has a means by which the mobile terminal can distinguish different cells on the downlink. The MT distinguishes which cell (sector) is transmitting the signal on the downlink by looking at the scrambling code. There are a total of a total of 512 possible scrambling codes on the downlink.

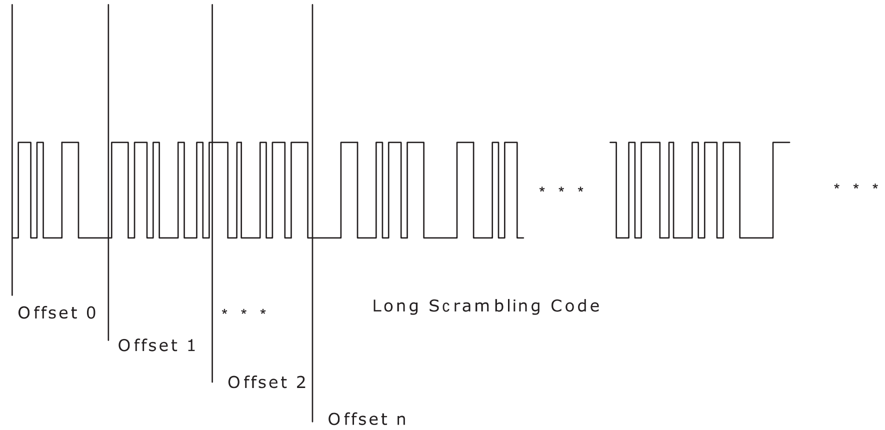
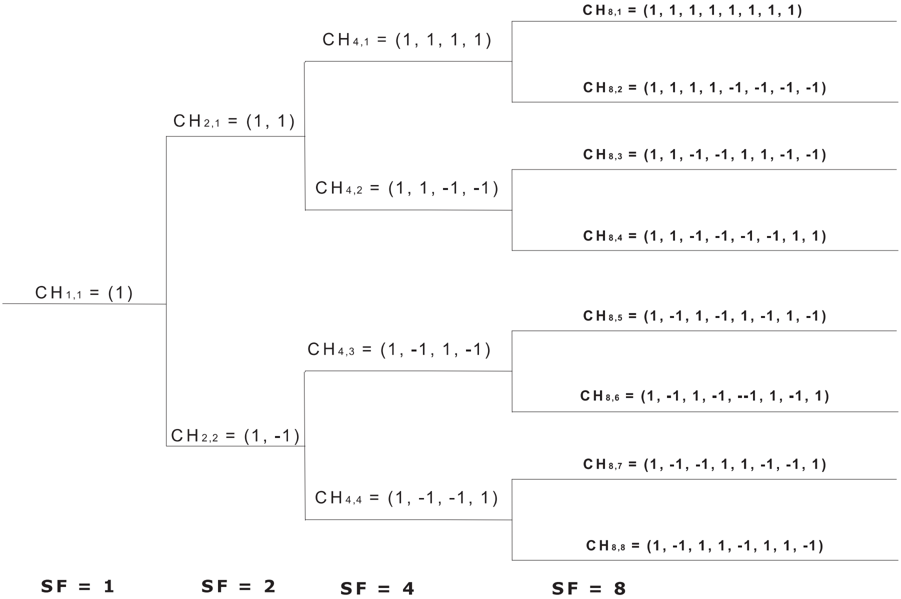


Figure 15.11 WCDMA scrambling codes with different offsets.



-Orthogonal Variable Spreading Factor (OVSF) code
-Provides channel separation in either direction
-Codes are orthogonal as long as within the spreading factor (SF)

Figure 15.12 WCDMA orthogonal variable spreading factor (OVSF).

Uplink Scrambling Codes WCDMA has a means by which the cell can distinguish different mobiles on the uplink. The cell distinguishes which MT is transmitting a signal on the uplink by looking at another set of scrambling codes. There are a total of 16.8 million possible scrambling codes on the uplink.

15.8 Typical Link Budget

The following is a typical UMTS uplink budget that will serve as a baseline example. On account of the power limitation of the mobile terminal or the UE, the uplink is considered the limiting link, and in that sense, serves as the minimum power requirement link. The link budget presented here has the following calculations:

1. Transmit power calculation;
2. Noise and interference calculation;
3. Margin and loss incorporation;

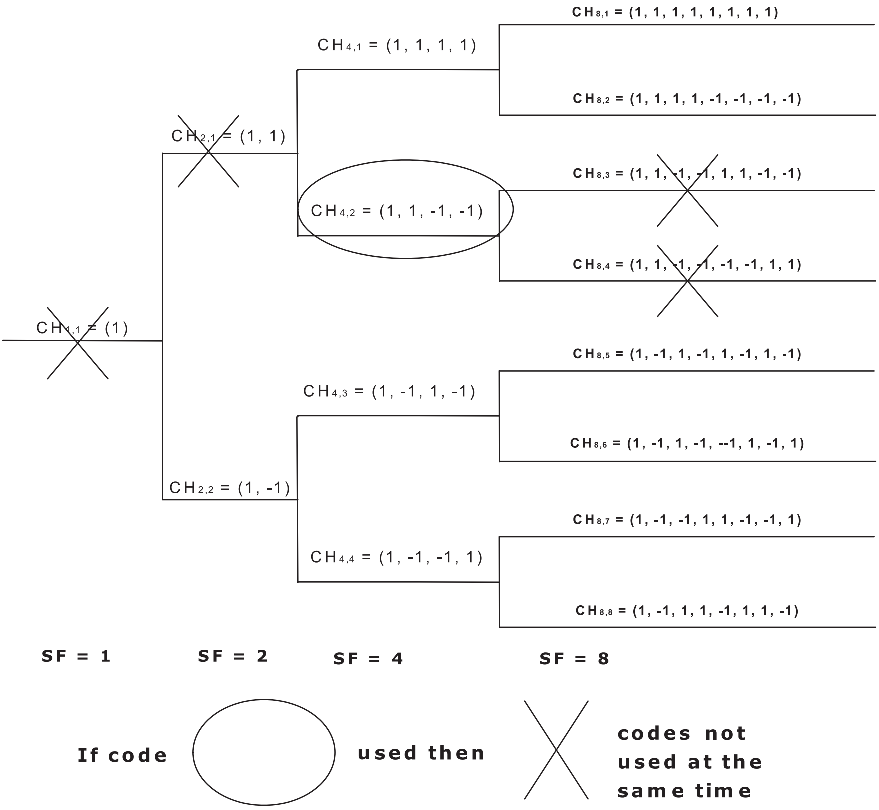


Figure 15.13 WCDMA orthogonal variable spreading factor (OVSF), showing codes used and not used to maintain orthogonality.

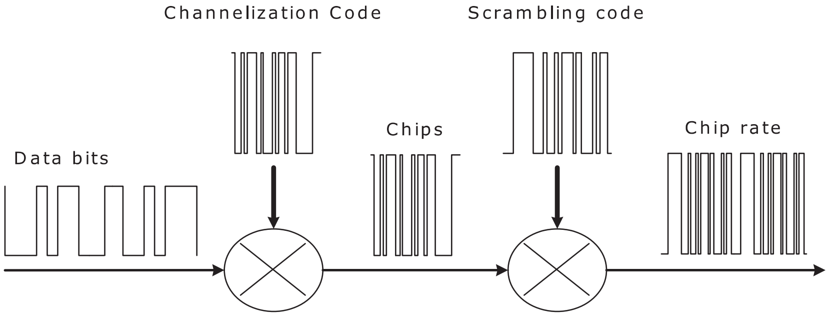


Figure 15.14 Channelization and scrambling codes applied to data bits one after the other.

Table 15.2
UMTS Link Budget

UMTS Uplink Link Budget	
Mobile power (dBm)	21
Antenna gain (dB)	4
Body loss (dB)	2
EIRP (dBm)	**19**
BTS noise power density (dBm/Hz)	–174
Noise figure (dB)	6
System Noise Floor (dBm/Hz) (= Noise Power Density + NF)	**–168**
Receiver Noise Power (dBm)	**–102.2**
Interference margin	3
Noise and Interference (dBm)	–99.2
Processing Gain (dB), 13K Vocoder (= 10log(3,840/13)	25
Required E_b/N_0 for speech (dB)	5
Receiver antenna gain (dBi)	17
Cable and connector loss	3
Fast fading margin (dB) slow moving mobile	4
Receiver Sensitivity (dBm) (= Noise + Interference – Processing Gain + Eb / No)	**–119.2**
Required IRL (dBm) (= Rxsens – Gant + Lcable)	**–133.2**
System Gain (dB) (= EIRP - Required IRL)	**152.2**
MAPL (dB) (= System Gain – Margin)	**148.2**
Cell Planning	
Coverage probability %	90
Log normal fading margin (dB)	7
Cost 231 model exponent	3.52
Indoor/in-vehicle loss (dB)	0
Soft-handover gain (dB)	3
Cell Edge Target Propagation Loss (dB) (=MAPL – Fading Margin + Soft–Handover Gain)	**144.2**
Okumura-Hata Cell Range (R km) (R) (from MAPL=137.4 + 35.2log(R); R = 10{(MAPL – 137.4)/35.2)}	**1.56**

4. Receiver sensitivity computation;
5. Path loss calculation;

6. Cell radius calculation that takes into account the random and probabilistic nature of the radio frequency (RF) signal.

Varying the fade margin from 0 to 7 dB will result in 50–90% coverage probabilities, respectively (assuming the industry accepted standard deviation of 8 dB). The most important outcome of this link budget is a determination of the coverage reach or range of the system, as manifested by two interrelated variables: maximum allowable path loss (MAPL) and cell radius.

15.9 Typical Capacity

15.9.1 HSDPA Capacity

High-speed downlink packet access (HSDPA) is said to deliver 14 Mbps under the following conditions. Here is how.

Using an OVSF of 16 (and chip rate of 3.84·106 Bps) and a 16 QAM modulation scheme, each channel will have 3.84·106/16 = 240 kbps.

If we set aside one channel for signaling, then the 15 channels deliver 15 × 240 kbps = 3.6 Mbps.

Modulation of 16 QAM will result in a factor of 4 (with each symbol containing 4 bits with 16 possible symbols). Then 3.6 Mbps·4 = 14 Mbps.

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16

Competing with and Complementing WiMAX

In this chapter, various key technologies that compete with as well as complement WiMAX are discussed. An example of those technologies that WiMAX complements and that are discussed here include SS7 and WiFi, whereas those wireless standards it competes with include LTE and UMB.

SS7 is discussed at a greater length on account of its usefulness and the versatility it provides to the communications world. Following that, in case of link failure within SS7, it is shown how WiMAX serves as wireless alternative [1]. WiMAX is also described as complementing WiFi in two configurations in which:

1. WiMAX serves as a backhauling mechanism from rooftop or tower top WiFi antennas—WiFi mesh application;
2. WiMAX cell sites serve as macro sites covering pedestrian and mobile subscribers, while residential, office, and commercials hotspots are covered by WiFi rooftop antennas—an overlay situation.

Next, LTE and UMB are presented here as contenders for dominance providing a similar service to that of WiMAX. By the same token, the fact that WiMAX can interface with IP services, including LTE and UMTS, is underscored.

16.1 Signaling System 7

Signaling system 7 (SS7) [1] is a set of telecommunications signaling protocols defined by the International Telecommunication Union (ITU), and it is used in the support of call service for both mobile and fixed applications. The signaling occurs out-of-band on dedicated channels, rather than in-band on voice channels, providing for faster call setup, more efficient use of voice circuits, the support of intelligent network (IN) services, which require signaling to network elements without voice trunks, and improved control over fraudulent network usage.

SS7 is the world's largest versatile and robust data network that links landline, cellular, and long-distance networks nationwide and internationally. As such, SS7 interconnects thousands of phone company providers into one common signaling network.

SS7 is based upon packet-switching technology. SS7 packets (or messages) are used to convey signaling information from an originating point to a destination point, through multiple switching nodes in the network. The SS7 messages contain addressing and control information that is used to select the routing of signaling information through the network, perform management functions, provide high reliability, establish, maintain, and tear down calls, and invoke transaction-based mechanisms in support of sophisticated applications. A transaction is simply a controlled exchange of information based on a query or command and the response to that query or command. The SS7 transaction-operation mechanisms are used to query databases and invoke functions at remote points throughout the network. These mechanisms also support the delivery of status information and results to those database queries and invoked functions.

Since SS7 has so many advantages and provides a standard transaction-based protocol mechanism, it is ideal for performing the operations required to provide the mobility management function in the mobile telecommunications network.

16.1.1 ANSI-41

One of the important uses of SS7, insofar as mobile communication is concerned, is the carrying of the ANSI-41 signaling protocol. Used on AMPS, TDMA, and CDMA, ANSI-41 (IS-41) is a standard for identifying and authenticating users and routing calls on mobile phone networks. The standard also defines how users are identified and calls are routed when roaming across different networks.

ANSI-41 signaling protocol provides, for example, operations for subscriber mobility in the mobile telecommunications network, and it is transported by SS7 protocol. While other protocols, such as CCITT X.25

packet-switching protocols, can be used to carry ANSI-41, SS7 has proved to be more powerful and robust. ANSI-41, in order to provide mobility and roaming management, relies on transaction capability application (TCAP) and signaling connection control part (SCCP) protocols, which are particular to SS7 networks and are not found in cellular network. ANSI-41 then provides the messages transactions necessary to register and cancel registration in various databases, while TCAP and SCCP provide the routing and transport of these messages. Without ANSI-41, users would have to call ahead and arrange for a special roaming number.

Refer to Figures 16.1 and 16.2, which show SS7 protocol.

16.1.2 Protocol

Refer to Figures 16.1 and 16.2. Levels 1, 2, and 3, equivalent to the OSI model, are combined into one entity in SS7, the message transfer part (MTP). MTP

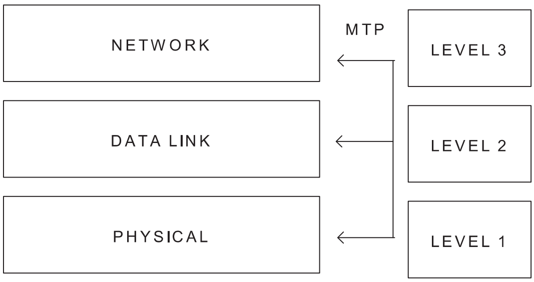

Figure 16.1 SS7 message transfer part.

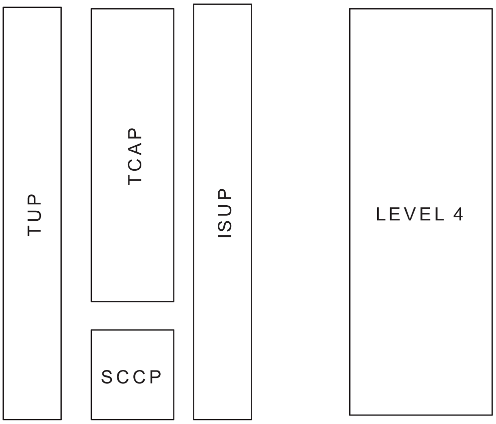

Figure 16.2 SS7 application.

provides the rest of the levels with node-to-node transmission, including basic error detection and correction schemes and message sequencing, routing, discrimination, and distribution functions within a node. More particularly, corresponding to level 1, the physical layer is responsible for converting the digital data into a bitstream for transmission over the network. Corresponding to level 2, the data-link layer provides the SS7 network with error detection/correction and sequenced delivery of all SS7 messages. Corresponding to level 3, the network layer provides three functions: routing, message discrimination, and distribution.

User part, as part of level 4 in the SS7 network, consists of several different protocols called user parts and application parts. For basic telephone call connect and disconnect, the telephone user part (TUP) or ISDN user part (ISUP) protocols are used. To access network databases, the transaction capabilities application part (TCAP) is used.

TCAP supports the functions required to connect to an external database, perform a query of the database, and retrieve information. The information or data retrieved thus is sent back in the form of a TCAP message to the signaling point that requested it. It also supports the remote control of other entities in the network, in which a network switch can invoke a feature or a function in another network switch by sending a cap message from one entity to another. Last but not least, TCAP also supports the messages for the operation, maintenance, and administration part (OMAP).

The signal connection control part (SCCP) provides connectionless and connection-oriented network services and global title translation (GTT) capabilities above MTP level 3. A global title is an address (for example, a dialed 800 number, calling card number, or mobile subscriber identification number) that is translated by SCCP into a destination-point code and subsystem number. A subsystem number uniquely identifies an application at the destination signaling point. SCCP is used as the transport layer for TCAP-based services.

When a database transaction is requested, MTP is accompanied by another higher level protocol, SCCP. SCCP provides the addressing necessary to route a message to the correct database.

The ISDN user part (ISUP) and message transfer part (MTP), which are made up of the three lower layers, physical, data link, and network, are used in the ANSI-41 network to connect cellular calls to the public switched-telephone network and to connect cellular circuits from the mobile switching center (MSC) to the base station (BS). ISUP is the protocol used to set up and tear down telephone connections between end offices. This protocol is derived from the telephone user part (TUP), but offers the added benefit of supporting intelligent networking functions and ISDN services. ISUP is used throughout the United States.

ANSI-41 uses an MSC for connecting to other networks and databases. Little has changed functionally within the MSC, other than the fact that the data stored in the HLR can now be shared with other MSCs across the SS7 network. This requires a Transport Protocol (TCAP) to move the data through the SS7 network.

16.1.3 ISUP

Call setup and teardown function, provided in the ISUP application is the centerpiece and core application of an SS7 system. The following is a brief description of the call setup and teardown of ISDN using SS7 signaling.

The ISUP protocol was intended for use with digital subscriber interfaces, such as ISDN. When an ISDN subscriber picks up the phone, a call setup is initiated. When the signal gets to the ISDN switch (local exchange), it is converted to an SS7 IAM (initial address message). The IAM contains the same information as the SETUP message. In fact, there is direct mapping from the ISDN protocol to the SS7, making them compatible interworking protocols. Once the IAM reaches the distance exchange, the signal is converted back to an ISDN SETUP message, and the called party returns an ISDN ALERTING message. This alerting message indicates that all the addressing signals have been received, and the ISDN terminal (telephone) is now ringing. No ringing generator is sent from the local exchange, because ISDN is digital. The ringing is generated at the telephone device itself, based on the receipt of the SETUP message. When the exchange receives the ALERTING message, it is converted into the SS7 ACM, which indicates the called party is being signaled. The ACM at the originating exchange is converted into an ALERTING message. When the called party answers, the CONNECT message (SS7 equivalent message ANM) is sent in the backward direction. The call release is done in a similar manner.

16.1.4 SS7 Network

Refer to Figure 16.3. Connected to telephones on one side and STPs on the other, SSPs make possible the initiation, managing, and release of voice calls. SSPs are also capable of sending a query message to a centralized database, or SCP, to determine how to route a call. The SSPs are also mutually interconnected for the purpose of voice trunk linking.

Signal transfer point (STP) routes each incoming message to an outgoing signaling link, based on routing information contained in the SS7 message. STP can be regarded as a router with a redundantly connected network for robustness and fail-safe operation.

Service control point (SCP), connected to STPs, provide the database repository.

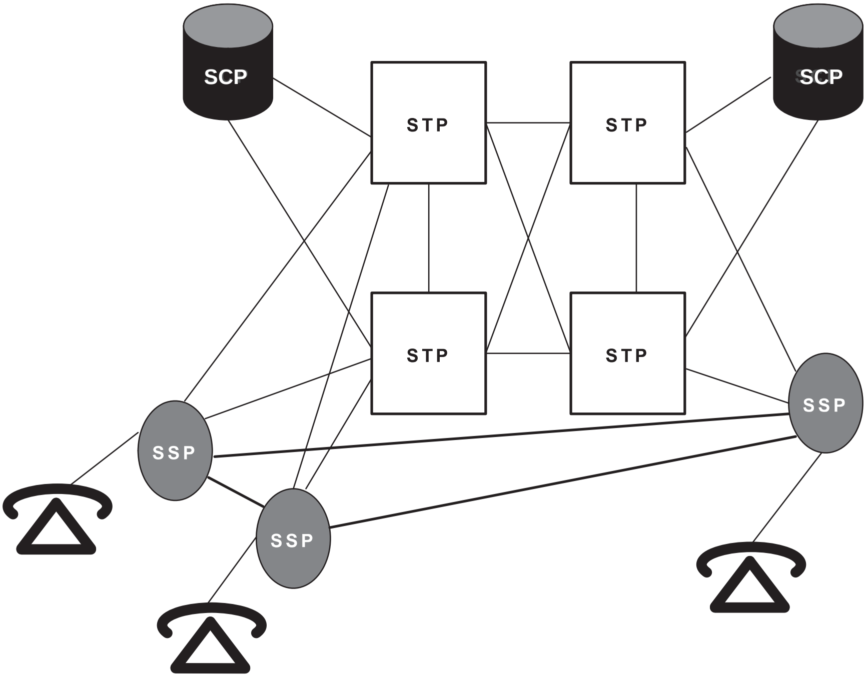

Figure 16.3 Typical SS7 architecture.

16.2 Wireless Failover Alternative for SS7

Refer to Figure 16.4. It is shown in the figure that, in the presence of a link failure between a switch (SSP) and STP, the wireless link enabled by WiMAX takes over.

The mediation device, when sensing the link failure, can reconstruct the protocol similar to the one going through the wireline and ship the data wirelessly.

While the patent applicants presented two alternatives—one with WiFi and the other with WiMAX—what is shown highlights the capability of same to complement SS7, providing a failsafe operation.

16.3 WiMAX Backhaul for WiFi Mesh

WiFi is a short-range alternative, measured in tens of meters, to WiMAX's long range, measured in many kilometers. WiFi is low cost and easily deployable technology, while WiMAX is highly scalable with the capability to handle mobile handoff, management, and MIMO smart antenna system functions, as well as guaranteed QoS.

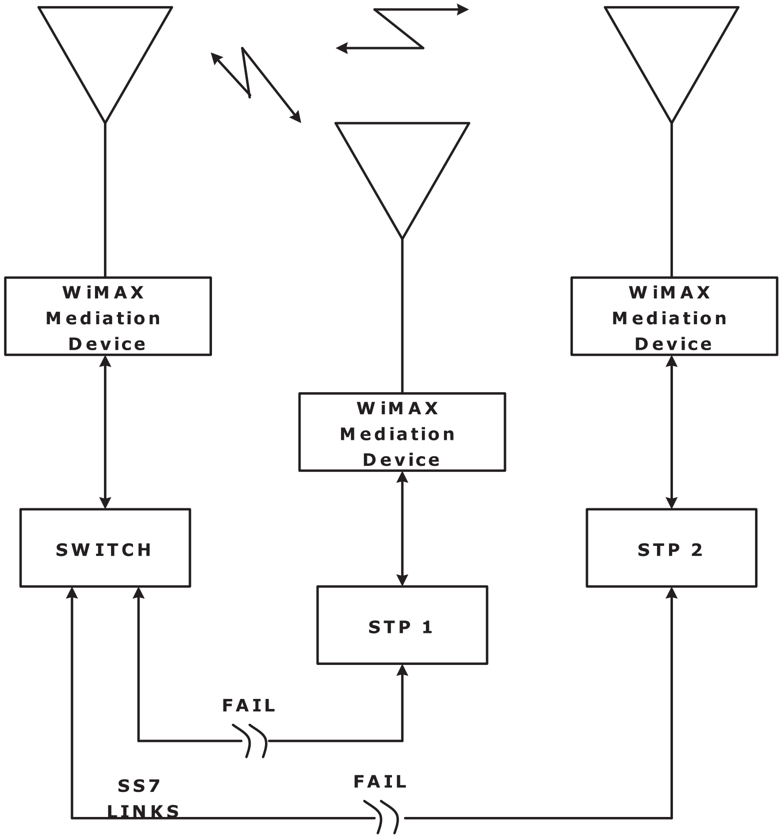

Figure 16.4 SS7 failover wireless link. (*From:* [2].)

WiMAX in WiFi Mesh Application Mesh networking, as it relates to WiFi, is a means of extending the range of traditional WLANs. In a mesh configuration, nodes are interconnected to provide a way of routing data, voice, and instruction between the various nodes. Self-healing, meshing also allows a means continuous reconfiguration around blocked paths, so that node-to-node transmission is maintained and optimized. Mobile ad hoc (wireless devices communicating in peer-to-peer fashion without a central access point) is closely related to mesh networking. Refer to Figure 16.5, which shows WiFi mesh, in which the pole-mounted access points (APs) are interconnected in a mesh fashion, and several of those APs are then connected to a roof-mounted or tower-mounted cell to supply the capacity-injecting layer. From the cell site back to the Internet, on account of its range and versatility, a WiMAX point-to-point backhauling network is used.

A combined WiFi mesh and WiMAX deployment, as shown in Figure 16.5, offers a more cost-effective solution than a sole WiFi directional-antenna

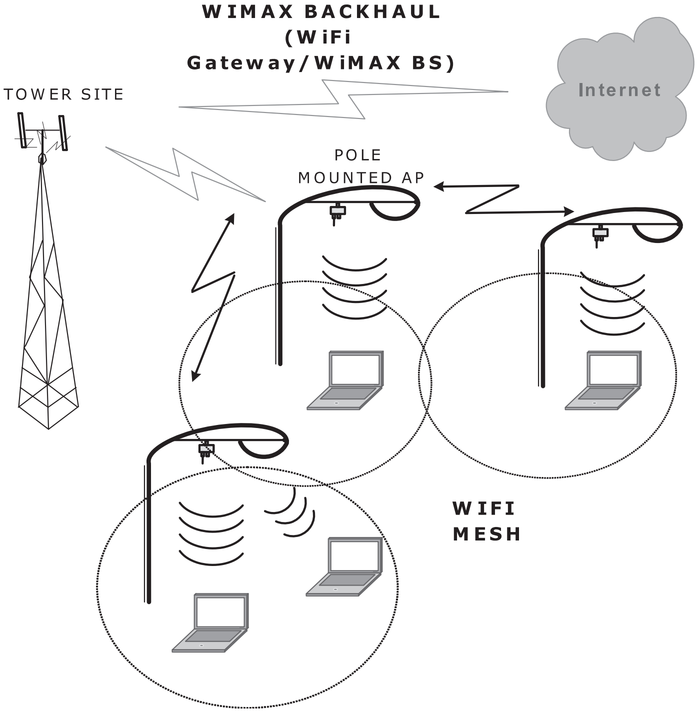

Figure 16.5 Using WiMAX to backhaul WiFi mesh.

deployment or a WiFi mesh network with a wired backhaul for wireless Internet service providers (WISPs) that want to extend the LAN or cover the last mile. With the emergence of WiMAX in the near future, a deployment that combines the two technologies can be constructed to take advantage of the strengths of both WiFi and WiMAX. A combination of WiMAX and WiFi mesh-network topology provides the best solution for this situation. WiMAX can be used to aggregate the community centers [3]. WiMAX extends the reach of broadband, while the proprietary WiFi mesh network available today can provide mobile client access throughout the community centers and park. As dual-mode WiFi and WiMAX cells are introduced into high-capacity network centers in licensed or unlicensed bands, the WiMAX cells will interoperate seamlessly with existing WiFi cells, always selecting the best path for delivering maximum user throughput end-to-end.

16.4 WiMAX Complementing WiFi Hotspot

Refer to Figure 16.6 showing WiFi and WiMAX complementing each other. WiFi serves residential, work, or commercial hotspots through dedicated

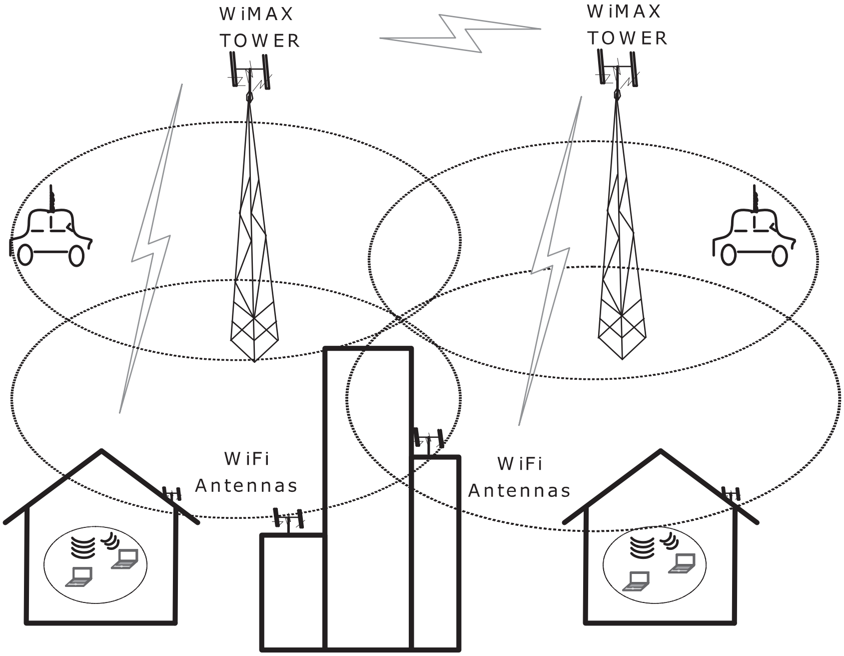

Figure 16.6 WiFi and WiMAX working together. WiFi is serving hotspots while WiMAX is serving at the macro level, including mobile subscribers.

high-gain antennas. The signals are then fed back to the WiMAX tower, from which it is backhauled to the destination. WiMAX also directly serves pedestrian and mobile subscribers at the macro level.

16.5 Working with IP-Based Technologies

From the foregoing discussion, it is clear, with the future pointing toward VOIP and Internet applications, an all-IP design is the best alternative. UMTS, HSPA, and LTE are all IP-based technologies, but they are encumbered with numerous migration and backward-compatibility requirements. Clearly, WiMAX has tremendous advantages from a few vantage points:

1. The architecture is simple.
2. The protocol implementation is clean, since it starts from scratch—IP.
3. The timing is correct—with the major players and investors and completion of the standard coming together.
4. The momentum of the acceptance of the technology around the world is converging.

5. A number of actual networks are increasingly going commercial around the world, not to mention the number of successful trials with promising results. ZTE, the biggest Chinese telecommunication enterprise and equipment provider, deployed Colombia's first mobile WiMAX network on August 25, 2008.

The two areas needing further evaluation, at the time of this writing, as far as WiMAX technology, are VoIP and mobility applications. Even these, it is believed, will be resolved in time.

The other advantageous aspect of WiMAX is certification. The WiMAX certification allows vendors with 802.16d products to sell their equipment as WiMAX certified, thus ensuring a level of interoperability with other certified products, as long as they fit the same profile. On the other hand, LTE is still on the drawing board and LTE specifications are yet to be certified. LTE is expected to be commercially available no earlier than 2011.

LTE, with all its promises, and UMB, with its supposed advantages, will not be available soon enough. In wireless technology, as dynamic as it is, timing is everything.

16.6 Versatility of WiMAX Network and Protocol

Refer to Figures 16.7 and 16.8, which show the WiMAX reference design model and protocol, respectively.

The design principles that WiMAX espouses include the following;

1. The architecture shall be decomposed into functions and well-defined reference points between functional entities for multivendor interoperability.
2. The architecture shall provide modularity and flexibility in deployment. Multiple types of decomposition topologies may coexist, such as distributed, centralized, and hybrid.
3. The architecture shall support fixed, nomadic, portable, and mobile operation and an evolution path to full mobility;
4. The architecture shall support the decomposition of access network and connectivity network; access network is radio-agnostic and connectivity network provides IP connectivity.
5. The architecture shall support sharing of the network with a variety of business models:

    a. A network access provider (NAP) owns the network and operates it.

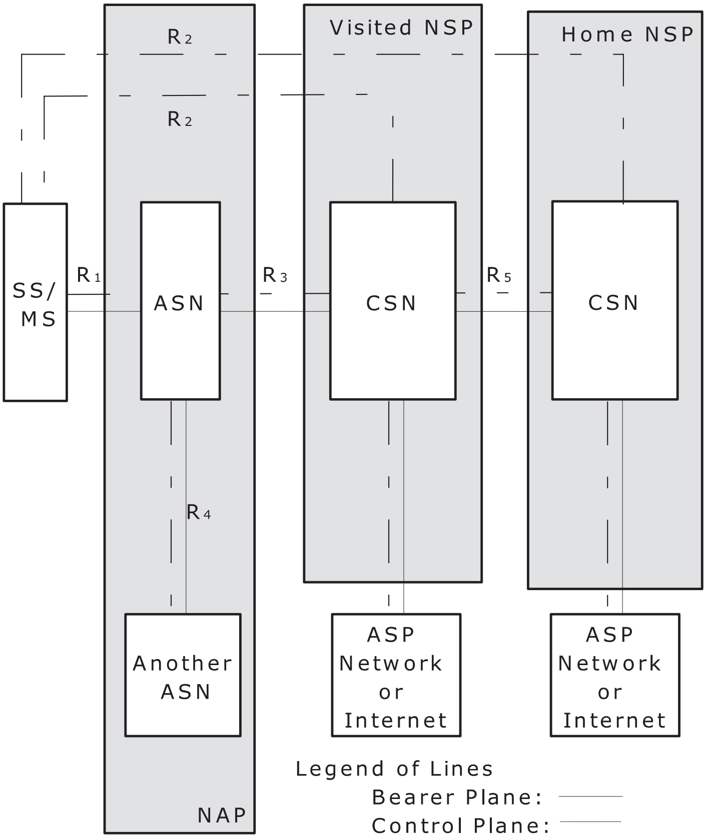

Figure 16.7 WiMAX architecture. (*From:* [4]. © 2007 WiChorus, Inc. Reprinted with permission.)

b. A network service provider (NSP) owns the subscriber and provides service. NSPs share the NAP or an NSP uses multiple NAPs.

c. An application service provider (ASP) provides application services.

6. The architecture shall support internetworking with 3GPP, 3GPP2, WiFi, and wireline networks, using IETF protocols.

Figure 16.7 shows the WiMAX network reference model (NRM). The figure illustrates the reference points and functional entities. The NRM is composed of three logical parts: mobile stations, the access service network (ASN), which is owned by NAPs, and the connectivity service network (CSN), which is owned by NSPs. The reference points, as seen in Figure 16.7, are conceptual links that connect the two functional entities. The reference points are:

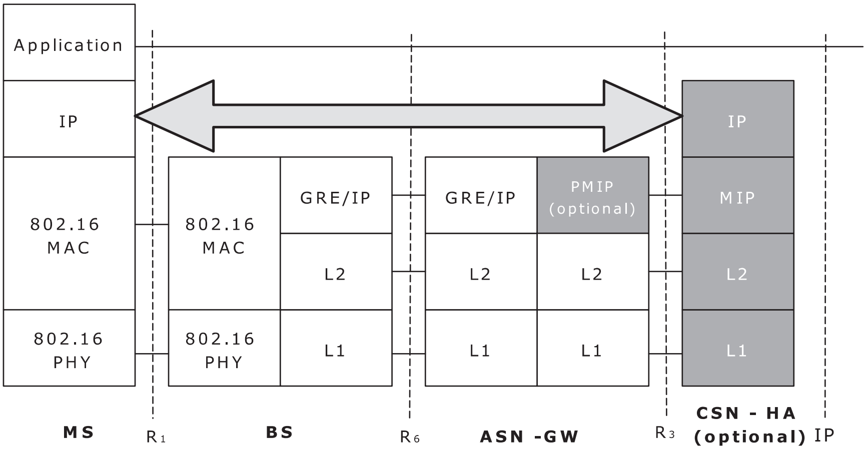

Figure 16.8 WiMAX protocol.

- R1: Reference point between MS and BS: implements IEEE 802.16e-2005;
- R2: Reference point between MS and ASN-GW or CSN: logical interface used for authentication, authorization, IP-host configuration, and mobility management;
- R3: Reference point between ASN and CSN: supports AAA, policy enforcement, and mobility-management capabilities and implements tunnel between ASN and CSN;
- R4: Reference point between ASN and ASN: used for MS mobility across ASNs;
- R5: Reference point between CSN and CSN: used for internetworking between home and visited network;
- R6: Reference point between BS and ASN: implements intra-ASN tunnels and used for control plane signaling;
- R7: Reference point between data and control plane in ASN-GW: used for coordination between data and control plane in ASN-GW;
- R8: Reference point between BS and BS: used for fast and seamless handover.

The access service network (ASN) provides a means to connect mobile subscribers using the OFDMA air link to the IP backbone with session continuity. ASN comprises base stations (BS) and access gateways named ASNGW. The interface between the ASN and mobile subscriber is through BS with the IEEE

802.16e-2005 (IEEE 802.16-2004 for some previous fixed configurations) standard.

It is clear to see from Figure 16.8, which shows the WiMAX protocol, that the implementation is clean, simple, and easy to implement. There are few protocols and few device requirements. The third layer, the routing layer, is an IP-layer end-to-end. The first two layers, the physical and data-link layers, are the 802.16 standard defining the air interface and the MAC layer, respectively. The WiMAX architecture can be used to support both the IP and the Ethernet packet. The IP packet may be transported using the IP convergence sublayer (IP-CS) over IEEE 802.16e or by using the Ethernet convergence sublayer (ETH-CS) over IEEE 802.16e. Within the ASN, the Ethernet or IP packets may be either routed or bridged. Routing over the ASN may be done using IP-in-IP encapsulation protocols, such as GRE (generic routing encapsulation).

16.7 Competing Against WiMAX

16.7.1 Overview

Going to the next advanced generation, the technology field is replete with competition. One might ask, "What is the impetus for going to the next generation of technology?" There is an existential need to go to the next generation, in this case the fourth generation (4G), on account of increasing demand in the areas of not only just Internet connection for emails and text messages, but also higher demand for multimedia applications, such as streaming audio and video. At this time, we play, work, pay our bills, shop, and get our news directly through the Internet. Unlike in the past, we would like to do these tasks from anywhere, while in transit, and preferably at an efficient rate and in a secured environment. Hence, mobility, quality of service, and secured communications also play a greater part in creating more demands. In the United States, where landline infrastructure was built early on, the demand for wireless, nomadic, or mobile service to exchange data is not as strong as it is in the rest of the world. Nevertheless, the demand is building momentum here, too. And those who tasted the luxury of whatever services from anywhere cannot go back easily to tethered and slow services.

The demands for services such as voice over IP (VoIP), which require especially higher bandwidth (and not to mention its intolerance of delays) can be comfortably met in the improvements incorporated in 4G. The 4G scheme goes all-out IP, for starters. Secondly, nearly all existing technologies aspiring to provide more efficient and higher data rates use an improved modulation scheme, more particularly, OFDM. In addition, higher bandwidth is used. Compared to legacy systems (2G and 3G), 4G uses much higher bandwidth to deliver a high data rate. However, in order to be backward compatible, 4G scales bandwidths, starting, say, at 1.25 MHz, in small increments, all the way to 20 MHz. The other huge advantage 4G enjoys over the previous generation, in order to deliver a high data rate, is the antenna technology named multiple-input-multiple-output (MIMO). In this antenna technology, the adverse effect of multipath is taken advantage of to improve reception. In an environment in which there is significant reflection, such as in a dense urban area, for example, MIMO comes to the rescue. So in effect, scalability, high bandwidth, OFDM modulation scheme, all-IP technology, coupled with MIMO antenna technology, deliver the high data rate demanded in 4G.

There are a few technologies on the horizon willing to meet the challenges of 4G. One is WiMAX, the subject of this book. WiMAX is a purely IP-based technology, with a standard written from scratch by the WiMAX Forum. The other is LTE, a burgeoning technology along the GSM evolution. LTE comes after a long line of constant migration technologies: the evolution was from GSM → GPRS → EDGE → GERAN → UMTS → HSPA → LTE.

The LTE standard was developed by the Third Generation Partnership Project (3GPP), which is also responsible for the GSM and the evolution of standards. LTE is in many ways similar to WiMAX (OFDM, turbo coding, and MIMO, for example). In some aspects it is better, but in some other aspects, it is worse. The expected time of deployment for LTE is no earlier than 2011. For a detailed technology roadmap, refer to [5].

The following evolution occurred on the CDMA side: IS-95 → CDMA2000 → evolution data option (EVDO) → Revision A, B, or C and → now, albeit a big shift in CDMA technology tradition, is ultrawideband (UWB), which is being developed by Qualcomm.

The Third Generation Partnership Project 2 (3GPP2) developed EVDO Rev C, as well as the CDMA standards. An IEEE working group is working on a standard for UWB.

Designed to elevate the performance of the 3GPP2 family of standards for wider bandwidths, UMB's objective is to be a ground-up, real-time application that delivers a mix of real-time VoIP, video, and Web traffic, for both consumer and corporate clients at high data rates for mobile and fixed environments. UMB is also projected to be low latency with rapidly maturing standards using OFDMA bandwidth ranges of 1.25 up to 20 MHz and leveraging advanced antenna techniques such as MIMO and beamforming.

A technology powerhouse based in San Diego, California, Qualcomm is advancing UMB now. In its heydays, in the 1990s, Qualcomm introduced CDMA, rather successfully, for commercial consumption. It has been 10 years now, and that technology is fast becoming obsolete, including CDMA2000 and its EVDO improvements. Seeing the writing on the wall, Qualcomm has been looking for newer technology, mostly through acquisition, to meet the ever-changing and more efficient demand. Not too long ago, Qualcomm acquired Flarion, which pioneered in fast low-latency access with seamless handoff-orthogonal frequency division multiplexing (FLASH-OFDM), with a capacity at least three times as much as CDMAs. While WiMAX is a better alternative than FLASH-OFDM from a performance standpoint, Qualcomm is said to own patents in both technologies.

16.7.2 Long-Term Evolution (LTE)

An entirely new radio platform technology and a GSM evolutionary path beyond 3G, long-term evolution (LTE) is a standard under development by 3GPP. The GSM evolutionary path of LTE is beyond 3G, and it is widely considered the 4G for GSM/WCDMA networks, following EDGE, UMTS/WCDMA, and HSPA (HSDPA and HSUPA combined).

LTE achieves high peak data rates in wide spectrum bandwidth as a result of using OFDMA. See Table 16.1 for LTE's specifications. It is believed that LTE technology will serve as a foundation for the GSM evolution path of 4G.

The main benefits of LTE can be summed up as:

- IP, packet-based network that lowers latency and cost;
- Flexibility of deployment spectrum as well as the scalability of bandwidth, which allows operators to tailor their network deployment strategies to fit their available spectrum resources.

Refer to Figure 16.9 [6] displaying the general architecture of LTE. The E-UTRAN consist of eNBs that are interconnected with each other by the X2 interface. Each eNB is connected to the evolved packet core (EPC) network by the S1 interface. On the user plane, the S1 interface terminates the serving gateway (S-GW), and, on the signaling plane, the S1 interface terminates the mobility management entity (MME).The eNBs are terminating points for the control

Table 16.1
LTE Specifications

Downlink (Data rate in 20 MHz)	100 Mps
Uplink (Data rate in 20 MHz)	50
Scalable Bandwidth (MHz)	1, 1.25, 2.5, 5, 10, 15, 20
Spectrum Efficiency	2 or 4 times that of HSPA
Latency (between UE and BS)	10 ms
Latency (Inactive to Active)	100 ms
Duplexing Scheme	FDD and TDD

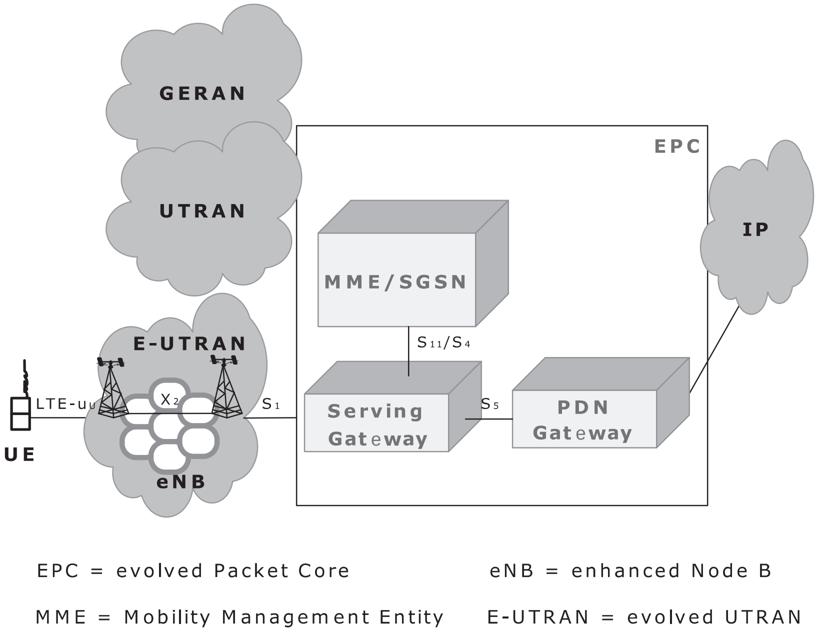

Figure 16.9 LTE architecture. (*From:* [6]. © 2007 3GPP. TSs and TRs property of ARIB, ATIS, CCSA, ETSI, TTA, and TTC. Reprinted with permission.)

and user planes toward the UEs in the evolved UTRA. The E-UTRAN's overall architecture is described in technical specifications 36.300 and 36.401.

16.7.3 Comments and Opinions on UMB and WiMAX

Qualcomm, not only as the proponent, but also as the developer of ultramobile broadband (UMB), maintains [7]:

> …UMB is the next evolutionary step within third generation partnership project 2[1] (3GPP2) using OFDMA and designed for robust mobile operations in frequency reuse-1. It has been designed for mobility support as well as a rich mix of VoIP and data services. Mobile WiMAX, which is based on 802.16e, is the evolutionary step for fixed WiMAX based on 802.16d. Even though Mobile WiMAX does use OFDMA, it is not suitable for robust frequency reuse-1 operation in a mobile environment. Performance simulations show that mobile WiMAX offers low spectral efficiency compared

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[1] 3GPP2 is the standardization group for CDMA2000, the set of 3G standards based on earlier 2G CDMA technology, not to be confused with 3GPP, which specifies standards for the 3G technology of UMTS.

> with UMB under similar assumptions. In addition, support for real-time applications such as VoIP significantly reduces WiMAX network capacity, thus resulting in higher capital and operating costs.

To counter the above statement by Qualcomm, comments by ZTE executive Shen that the CDMA operator will choose LTE [8][2]:

> …China Telecom, which recently bought the Unicom's CDMA business, had decided to ditch UMB in favor of LTE. Chairman Wang Xiaochu says the company will initially upgrade the CDMA network to EV-DO Rev A and eventually migrate to LTE. [Shen added,] …China Telecom is just one of a string of CDMA carriers around the world, such as Verizon Wireless, which will begin testing LTE next year.
>
> This is a global trend: CDMA is shrinking and the economies of scale between CDMA and GSM cannot be compared, [he told the *Show Daily*] CDMA as a 2G technology will continue for a while to support voice service, but at the end of the day the majority of [EV-DO] operators will eventually go for LTE, not UMB.
>
> While WiMAX is gaining traction, Shen suggested, the opportunity for WiMAX remains modest because it is more "vendor-driven" and does not yet support full mobility. "WiMAX will be a technology for niche markets, as such fixed replacement, and not likely to be widely adopted," he said.

While these statements might have been true at the time of the comment, mobility trials and actual deployment and commitment on behalf of WiMAX technology are proving to be very promising. At the time of this writing, over 42 countries around the world have deployed WiMAX successfully. With Pakistan leading the pack with the largest functional WiMAX network, from Ethiopia to Estonia, to North and South America, to Malaysia to South Korea, to Bolivia to Congo, to Brazil to Colombia, the technology is generating a huge momentum with an eye toward reaching up to 140 million subscribers by 2010. While cellular technologies went through years of development and improvements to get to where they are now, WiMAX will be a high-speed all-IP core network right from the start.

16.8 The Competition for Dominance

The pros and cons of LTE and WiMAX are as follows.

LTE as it stands has a fundamental and existential problems. It is not here yet, at least not until 2010. LTE, on paper, is said to be better for FDD

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[2] Qualcomm recently announced that it has stopped UMB development and joined the LTE bandwagon.

implementation, while WiMAX is better for TDD. WiMAX has a newer up-and-coming technology as well—IEEE 802.16m, a follow-on to the 802.16e standard and backward compatible and capable of FDD and TDD modes as well as a 350-km/hr mobility application.

LTE, as yet to see any deployment, promises to be another extension of a long evolution of GSM to UMTS. On the other hand WiMAX, as a standalone, pure IP-based technology and unencumbered with backward compatibility constraints, is already deployed with full-blown mobility application coming to fruition.

Perhaps a significant advantage that LTE will enjoy, when it does become reality, is leveraging on existing more than 2.5 billion subscribers in the GSM/UMTS evolution of technologies, thus tipping the balance of the economies of scale. This means that infrastructure and service costs can be significantly lower on account of mass production. As a case in point, in the United States, both AT&T and Verizon have chosen LTE as their vehicle for their 4G migration.

However, one of the disadvantages for LTE is that it is not a natural migration from GSM. It is believed that it would require erasing more than 90% of existing capital expenses. Conversely, if WiMAX continues to build on its deployment successes around the world (including improved data rates, mobility, and VoIP performances), this can translate into significant increases in its customer base. At the time of this writing, WiMAX has 100 global commercial deployments with 500 vendors committed to build WiMAX products and services, which gives it a tremendous deployment cost advantage; hence, WiMAX's economies of scale are growing too, though not at the LTE's rate. In addition, in what is believed to be a major milestone, on September 29, 2008, Sprint launched an XOHM WiMAX mobile broadband commercial service launch in Baltimore, Maryland, with many more cities to follow.

With these roaring successes and inherently clean, IP-based, and cost-effective technology, WiMAX will provide a compelling alternative to, if not outright better technology than, LTE.

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[7] Qualcomm, "Executive Summary of UMB and WiMAX Technical Comparison," January 2008, http://www.qualcomm.com/common/documents/white_papers/UMB_WIMAX.pdf.

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Appendix A
Erlang B

Table A.1
Erlang B

	Probability of Blocking									
No. of Trunked Channels	**0.001%**	**0.005%**	**0.010%**	**0.050%**	**0.100%**	**0.200%**	**0.300%**	**0.400%**	**0.500%**	**0.600%**
	Traffic Intensity in Erlangs									
1	0.0000	0.0001	0.0001	0.001	0.001	0.002	0.003	0.004	0.005	0.006
2	0.004	0.010	0.014	0.032	0.046	0.065	0.081	0.094	0.105	0.116
3	0.040	0.068	0.087	0.152	0.194	0.249	0.289	0.321	0.349	0.374
4	0.129	0.196	0.235	0.362	0.439	0.535	0.602	0.656	0.701	0.741
5	0.276	0.389	0.452	0.649	0.762	0.900	0.994	1.069	1.132	1.187
6	0.476	0.639	0.728	0.996	1.146	1.325	1.447	1.542	1.622	1.691
7	0.724	0.939	1.054	1.392	1.579	1.798	1.946	2.061	2.158	2.241
8	1.013	1.282	1.422	1.830	2.051	2.311	2.484	2.618	2.730	2.827
9	1.339	1.660	1.826	2.302	2.558	2.855	3.053	3.206	3.333	3.442
10	1.697	2.069	2.260	2.803	3.092	3.427	3.648	3.819	3.961	4.083
11	2.085	2.506	2.722	3.329	3.651	4.022	4.266	4.455	4.610	4.745
12	2.496	2.967	3.207	3.878	4.231	4.637	4.904	5.109	5.279	5.425
13	2.929	3.450	3.714	4.447	4.831	5.270	5.559	5.781	5.964	6.121
14	3.383	3.952	4.239	5.032	5.446	5.919	6.229	6.467	6.663	6.832
15	3.856	4.472	4.781	5.634	6.077	6.582	6.913	7.167	7.376	7.555
16	4.345	5.008	5.339	6.250	6.722	7.258	7.609	7.878	8.100	8.290
17	4.850	5.558	5.911	6.878	7.378	7.946	8.316	8.600	8.834	9.035
18	5.369	6.122	6.496	7.519	8.046	8.644	9.034	9.332	9.578	9.789

Table A.1 (continued)

	Probability of Blocking									
No. of Trunked Channels	**0.001%**	**0.005%**	**0.010%**	**0.050%**	**0.100%**	**0.200%**	**0.300%**	**0.400%**	**0.500%**	**0.600%**
19	5.902	6.698	7.093	8.170	8.724	9.352	9.761	10.073	10.331	10.552
20	6.446	7.285	7.701	8.831	9.412	10.068	10.496	10.823	11.092	11.322
21	7.002	7.883	8.319	9.501	10.108	10.793	11.239	11.580	11.860	12.100
22	7.568	8.493	8.946	10.180	10.812	11.525	11.989	12.344	12.635	12.885
23	8.144	9.110	9.583	10.868	11.524	12.265	12.746	13.114	13.416	13.676
24	8.730	9.735	10.227	11.562	12.243	13.011	13.510	13.891	14.204	14.472
25	9.324	10.369	10.880	12.264	12.969	13.763	14.279	14.673	14.997	15.274
26	9.927	11.010	11.540	12.972	13.701	14.522	15.054	15.461	15.795	16.081
27	10.537	11.659	12.207	13.686	14.439	15.285	15.835	16.254	16.598	16.893
28	11.154	12.314	12.880	14.406	15.182	16.054	16.620	17.051	17.406	17.709
29	11.779	12.976	13.560	15.132	15.930	16.828	17.410	17.853	18.218	18.530
30	12.417	13.644	14.246	15.863	16.684	17.606	18.204	18.660	19.034	19.355
31	13.054	14.318	14.937	16.599	17.442	18.389	19.002	19.470	19.854	20.183
32	13.697	14.998	15.633	17.340	18.205	19.176	19.805	20.284	20.678	21.015
33	14.346	15.682	16.335	18.085	18.972	19.966	20.611	21.102	21.505	21.850
34	15.001	16.372	17.041	18.835	19.743	20.761	21.421	21.923	22.336	22.689
35	15.660	17.067	17.752	19.589	20.517	21.559	22.234	22.748	23.169	23.531
36	16.325	17.766	18.468	20.347	21.296	22.361	23.050	23.575	24.006	24.376
37	16.995	18.470	19.188	21.108	22.078	23.166	23.870	24.406	24.846	25.223
38	17.669	19.178	19.911	21.873	22.864	23.974	24.692	25.240	25.689	26.074
39	18.348	19.890	20.640	22.642	23.652	24.785	25.518	26.076	26.534	26.926
40	19.031	20.606	21.372	23.414	24.444	25.599	26.346	26.915	27.382	27.782
41	19.718	21.326	22.107	24.189	25.239	26.416	27.177	27.756	28.232	28.640
42	20.409	22.049	22.846	24.967	26.037	27.235	28.010	28.600	29.085	29.500
43	21.104	22.776	23.587	25.748	26.837	28.057	28.846	29.447	29.940	30.362
44	21.803	23.507	24.333	26.532	27.641	28.882	29.684	30.295	30.797	31.227
45	22.505	24.240	25.081	27.319	28.447	29.708	30.525	31.146	31.656	32.093
46	23.211	24.977	25.833	28.109	29.255	30.538	31.367	31.999	32.517	32.962
47	23.921	25.717	26.587	28.901	30.066	31.369	32.212	32.854	33.381	33.832
48	24.633	26.460	27.344	29.696	30.879	32.203	33.059	33.711	34.246	34.704
49	25.349	27.206	28.104	30.493	31.694	33.039	33.908	34.570	35.113	35.578
50	26.067	27.954	28.867	31.292	32.512	33.876	34.759	35.431	35.982	36.454

Table A.2

Erlang B

	Probability of Blocking								
No. of Trunked Channels	0.70%	0.80%	0.90%	1.00%	2.00%	3.00%	5.00%	10.00%	20.00%
	Traffic Intensity in Erlangs								
1	0.007	0.008	0.009	0.010	0.020	0.031	0.053	0.111	0.250
2	0.126	0.135	0.144	0.153	0.223	0.282	0.381	0.595	1.000
3	0.397	0.418	0.437	0.455	0.602	0.715	0.899	1.271	1.930
4	0.777	0.810	0.841	0.869	1.092	1.259	1.525	2.045	2.945
5	1.236	1.281	1.322	1.361	1.657	1.875	2.219	2.881	4.010
6	1.753	1.809	1.861	1.909	2.276	2.543	2.960	3.758	5.109
7	2.315	2.382	2.444	2.501	2.935	3.250	3.738	4.666	6.230
8	2.913	2.990	3.062	3.128	3.627	3.987	4.543	5.597	7.369
9	3.540	3.627	3.708	3.783	4.345	4.748	5.370	6.546	8.522
10	4.191	4.289	4.378	4.461	5.084	5.529	6.216	7.511	9.685
11	4.864	4.971	5.069	5.160	5.842	6.328	7.076	8.487	10.857
12	5.554	5.671	5.777	5.876	6.615	7.141	7.950	9.474	12.036
13	6.261	6.386	6.501	6.607	7.402	7.967	8.835	10.470	13.222
14	6.981	7.116	7.238	7.352	8.200	8.804	9.730	11.473	14.413
15	7.714	7.857	7.987	8.108	9.010	9.650	10.633	12.484	15.608
16	8.458	8.609	8.747	8.875	9.828	10.505	11.544	13.500	16.807
17	9.212	9.371	9.517	9.652	10.656	11.368	12.461	14.522	18.010
18	9.975	10.143	10.296	10.437	11.491	12.238	13.385	15.548	19.216
19	10.747	10.922	11.082	11.230	12.333	13.115	14.315	16.579	20.424
20	11.526	11.709	11.876	12.031	13.182	13.997	15.249	17.613	21.635
21	12.312	12.503	12.677	12.838	14.036	14.885	16.189	18.651	22.848
22	13.105	13.303	13.484	13.651	14.896	15.778	17.132	19.692	24.064
23	13.904	14.110	14.297	14.470	15.761	16.675	18.080	20.737	25.281
24	14.709	14.922	15.116	15.295	16.631	17.577	19.031	21.784	26.499
25	15.519	15.739	15.939	16.125	17.505	18.483	19.985	22.833	27.720
26	16.334	16.561	16.768	16.959	18.383	19.392	20.943	23.885	28.941
27	17.153	17.387	17.601	17.797	19.265	20.305	21.904	24.939	30.164
28	17.977	18.218	18.438	18.640	20.150	21.221	22.867	25.995	31.388

Table A.2 (continued)

	Probability of Blocking								
No. of Trunked Channels	**0.70%**	**0.80%**	**0.90%**	**1.00%**	**2.00%**	**3.00%**	**5.00%**	**10.00%**	**20.00%**
29	18.805	19.053	19.279	19.487	21.039	22.140	23.833	27.053	32.614
30	19.637	19.891	20.123	20.337	21.932	23.062	24.802	28.113	33.840
31	20.473	20.734	20.972	21.191	22.827	23.987	25.773	29.174	35.067
32	21.312	21.580	21.823	22.048	23.725	24.914	26.746	30.237	36.295
33	22.155	22.429	22.678	22.909	24.626	25.844	27.721	31.301	37.524
34	23.001	23.281	23.536	23.772	25.529	26.776	28.698	32.367	38.754
35	23.849	24.136	24.397	24.638	26.435	27.711	29.677	33.434	39.985
36	24.701	24.994	25.261	25.507	27.343	28.647	30.657	34.503	41.216
37	25.556	25.854	26.127	26.378	28.254	29.585	31.640	35.572	42.448
38	26.413	26.718	26.996	27.252	29.166	30.526	32.624	36.643	43.680
39	27.272	27.583	27.867	28.129	30.081	31.468	33.609	37.715	44.913
40	28.134	28.451	28.741	29.007	30.997	32.412	34.596	38.787	46.147
41	28.999	29.322	29.616	29.888	31.916	33.357	35.584	39.861	47.381
42	29.866	30.194	30.494	30.771	32.836	34.305	36.574	40.936	48.616
43	30.734	31.069	31.374	31.656	33.758	35.253	37.565	42.011	49.851
44	31.605	31.946	32.256	32.543	34.682	36.203	38.557	43.088	51.086
45	32.478	32.824	33.140	33.432	35.607	37.155	39.550	44.165	52.322
46	33.353	33.705	34.026	34.322	36.534	38.108	40.545	45.243	53.559
47	34.230	34.587	34.913	35.215	37.462	39.062	41.540	46.322	54.796
48	35.108	35.471	35.803	36.109	38.392	40.018	42.537	47.401	56.033
49	35.988	36.357	36.694	37.004	39.323	40.975	43.534	48.481	57.270
50	36.870	37.245	37.586	37.901	40.255	41.933	44.533	49.562	58.508

Appendix B
Erlang C

Table B.1 shows the Erlang value corresponding to the probability a call will be delayed, shown as a percentage (%), for a given number of channels, *N*.

Table B.1
Erlang C

N	0.01%	0.05%	0.10%	0.50%	1%	2%	5%	10%	15%	20%	30%	40%
1	0.0001	0.0005	0.001	0.005	0.01	0.02	0.05	0.1	0.15	0.2	0.3	0.4
2	0.0142	0.0319	0.0452	0.1025	0.1465	0.2103	0.3422	0.5	0.6278	0.7403	0.939	1.117
3	0.086	0.149	0.1894	0.3339	0.4291	0.5545	0.7876	1.04	1.231	1.393	1.667	1.903
4	0.231	0.3533	0.4257	0.6641	0.81	0.9939	1.319	1.653	1.899	2.102	2.44	2.725
5	0.4428	0.6289	0.7342	1.065	1.259	1.497	1.905	2.313	2.607	2.847	3.241	3.569
6	0.711	0.9616	1.099	1.519	1.758	2.047	2.532	3.007	3.344	3.617	4.062	4.428
7	1.026	1.341	1.51	2.014	2.297	2.633	3.188	3.725	4.103	4.406	4.897	5.298
8	1.382	1.758	1.958	2.543	2.866	3.246	3.869	4.463	4.878	5.21	5.744	6.178
9	1.771	2.208	2.436	3.1	3.46	3.883	4.569	5.218	5.668	6.027	6.6	7.065
10	2.189	2.685	2.942	3.679	4.077	4.54	5.285	5.986	6.469	6.853	7.465	7.959
11	2.634	3.186	3.47	4.279	4.712	5.213	6.015	6.765	7.28	7.688	8.336	8.857
12	3.1	3.708	3.708	4.896	5.363	5.901	6.758	7.554	8.099	8.53	9.212	9.761
13	3.587	4.248	4.248	5.529	6.028	6.602	7.511	8.352	8.926	9.379	10.09	10.67
14	4.092	4.805	4.805	6.175	6.705	7.313	8.273	9.158	9.76	10.23	10.98	11.58

Table B.1 (continued)

N	0.01%	0.05%	0.10%	0.50%	1%	2%	5%	10%	15%	20%	30%	40%
15	4.614	5.377	5.377	6.833	7.394	8.035	9.044	9.97	10.6	11.09	11.87	12.49
16	5.15	5.962	5.962	7.502	8.093	8.766	9.822	10.79	11.44	11.96	12.77	13.41
17	5.699	6.56	6.56	8.182	8.801	9.505	10.61	11.61	12.29	12.83	13.66	14.33
18	6.261	7.169	7.169	8.871	9.518	10.25	11.4	12.44	13.15	13.7	14.56	15.25
19	6.835	7.788	7.788	9.568	10.24	11.01	12.2	13.28	14.01	14.58	15.47	16.18
20	7.419	8.417	8.417	10.27	10.97	11.77	13	14.12	14.87	15.45	16.37	17.1
21	8.013	9.055	9.055	10.99	11.71	12.53	13.81	14.96	15.73	16.34	17.28	18.03
22	8.616	9.702	9.702	11.7	12.46	13.3	14.62	15.81	16.6	17.22	18.19	18.96
23	9.228	10.36	10.36	12.43	13.21	14.08	15.43	16.65	17.47	18.11	19.1	19.89
24	9.848	11.02	11.02	13.16	13.96	14.86	16.25	17.51	18.35	19	20.02	20.82
25	10.48	11.69	11.69	13.9	14.72	15.65	17.08	18.36	19.22	19.89	20.93	21.76
26	11.11	12.36	12.36	14.64	15.49	16.44	17.91	19.22	20.1	20.79	21.85	22.69
27	11.75	13.04	13.04	15.38	16.26	17.23	18.74	20.08	20.98	21.68	22.77	23.63
28	12.4	13.73	13.73	16.14	17.03	18.03	19.57	20.95	21.87	22.58	23.69	24.57
29	13.05	14.42	14.42	16.89	17.81	18.83	20.41	21.82	22.75	23.48	24.61	25.5
30	13.71	15.12	15.12	17.65	18.59	19.64	21.25	22.68	23.64	24.38	25.54	26.44
31	14.38	15.82	15.82	18.42	19.37	20.45	22.09	23.56	24.53	25.29	26.46	27.38
32	15.05	16.53	16.53	19.18	20.16	21.26	22.93	24.43	25.42	26.19	27.39	28.33
33	15.72	17.24	17.24	19.95	20.95	22.07	23.78	25.3	26.32	27.1	28.31	29.27
34	16.4	17.95	17.95	20.73	21.75	22.89	24.63	26.18	27.21	28.01	29.24	30.21
35	17.09	18.67	18.67	21.51	22.55	23.71	25.48	27.06	28.11	28.92	30.17	31.16
36	17.78	19.39	19.39	22.29	23.35	24.53	26.34	27.94	29	29.83	31.1	32.1
37	18.47	20.12	20.12	23.07	24.15	25.36	27.19	28.82	29.9	30.74	32.03	33.05
38	19.17	20.85	20.85	23.86	24.96	26.18	28.05	29.71	30.8	31.65	32.97	34
39	19.87	21.59	21.59	24.65	25.77	27.01	28.91	30.59	31.71	32.57	33.9	34.94
40	20.58	22.33	22.33	25.44	26.58	27.84	29.77	31.48	32.61	33.48	34.83	35.89
41	21.28	23.07	23.07	26.23	27.39	28.68	30.63	32.37	33.51	34.4	35.77	36.84
42	22	23.81	23.81	27.03	28.21	29.51	31.5	33.26	34.42	35.32	36.7	37.79
43	22.71	24.56	24.56	27.83	29.02	30.35	32.36	34.15	35.33	36.23	37.64	38.74
44	23.43	25.31	25.31	28.63	29.84	31.19	33.23	35.04	36.23	37.15	38.58	39.69
45	24.15	26.06	26.06	29.44	30.67	32.03	34.1	35.93	37.14	38.07	39.51	40.64
46	24.88	26.82	26.82	30.24	31.49	32.87	34.97	36.83	38.05	39	40.45	41.59
47	25.6	27.57	27.57	31.05	32.32	33.72	35.84	37.72	38.96	39.92	41.39	42.54
48	26.34	28.33	28.33	31.86	33.14	34.56	36.72	38.62	39.87	40.84	42.33	43.5

Table B.1 (continued)

N	0.01%	0.05%	0.10%	0.50%	1%	2%	5%	10%	15%	20%	30%	40%
49	27.07	29.1	29.1	32.68	33.97	35.41	37.59	39.52	40.79	41.76	43.27	44.45
50	27.8	29.86	29.86	33.49	34.8	36.26	38.47	40.42	41.7	4?,69	44.21	45.4
51	28.54	30.63	30.63	34.31	35.64	37.11	39.35	41.32	42.61	43.61	45.15	46.36
52	29.28	31.4	31.4	35.12	36.47	37.97	40.23	42.22	43.53	44.54	46.1	47.31
53	30.03	32.17	32.17	35.94	37.31	38.82	41.1	43.12	44.44	45.47	47.04	48.27
54	30.77	32.95	32.95	36.76	38.15	39.67	41.99	44.02	45.36	46.39	47.98	49.22
55	31.52	33.72	33.72	37.59	38.99	40.53	42.87	44.93	46.28	47.32	48.93	50.18
56	32.27	34.5	34.5	38.41	39.83	41.39	43.75	45.83	47.2	48.25	49.87	51.13
57	33.03	35.28	35.28	39.24	40.67	42.25	44.64	46.74	48.12	49.18	50.82	52.09
58	33.78	36.06	36.06	40.07	41.51	43.11	45.52	47.64	49.04	50.11	51.76	53.05
59	34.54	36.85	36.85	40.9	42.36	43.97	46.41	48.55	49.96	51.04	52.71	54.01
60	35.3	37.63	37.63	41.73	43.2	44.83	47.29	49.46	50.88	51.97	53.65	54.96
61	36.06	38.42	38.42	42.56	44.05	45.7	48.18	50.37	51.8	52.9	54.6	55.92
62	36.82	39.21	39.21	43.39	44.9	46.56	49.07	51.27	52.72	53.83	55.55	56.88
63	37.59	40	40	44.23	45.75	47.43	49.96	52.18	53.64	54.77	56.49	57.84
64	38.35	40.8	40.8	45.06	46.6	48.3	50.85	53.1	54.57	55.7	57.44	58.8
65	39.12	41.59	41.59	45.9	47.45	49.16	51.74	54.01	55.49	56.63	58.39	59.76
66	39.89	42.39	42.39	46.74	48.3	50.03	52.64	54.92	56.42	57.57	59.34	60.72
67	40.66	43.18	43.18	47.58	49.16	50.9	53.53	55.83	57.34	58.5	60.29	61.68
68	41.44	43.98	43.98	48.42	50.01	51.77	54.42	56.75	58.27	59.44	61.24	62.64
69	42.21	44.78	44.78	49.26	50.87	52.65	55.32	57.66	59.2	60.37	62.19	63.6
70	42.99	45.58	45.58	50.1	51.73	53.52	56.21	58.57	60.12	61.31	63.14	64.56
71	43.77	46.39	46.39	50.95	52.59	54.39	57.11	59.49	61.05	62.25	64.09	65.52
72	44.55	47.19	47.19	51.79	53.45	55.27	58.01	60.41	61.98	63.18	65.04	66.48
73	45.33	48	48	52.64	54.31	56.14	58.9	61.32	62.91	64.12	65.99	67.44
74	46.11	48.81	48.81	53.49	55.17	57.02	59.8	62.24	63.84	65.06	66.94	68.4
75	46.9	49.61	49.61	54.34	56.03	57.9	60.7	63.16	64.76	66	67.89	69.37
76	47.68	50.42	50.42	55.19	56.89	58.78	61.6	64.07	65.69	66.94	68.85	70.33
77	48.47	51.23	51.23	56.04	57.76	59.65	62.5	64.99	66.63	67.88	69.8	71.29
78	49.26	52.05	52.05	56.89	58.62	60.53	63.4	65.91	67.56	68.82	70.75	72.25
79	50.05	52.86	52.86	57.74	59.49	61.41	64.3	66.83	68.49	69.76	71.7	73.22
80	50.84	53.68	53.68	58.6	60.36	62.3	65.21	67.75	69.42	70.7	72.66	74.18
81	51.63	54.49	54.49	59.45	61.22	63.18	66.11	68.67	70.35	71.64	73.61	75.14
82	52.43	55.31	55.31	60.3	62.09	64.06	67.01	69.59	71.28	72.58	74.57	76.11

Table B.1 (continued)

N	0.01%	0.05%	0.10%	0.50%	1%	2%	5%	10%	15%	20%	30%	40%
83	53.22	56.13	56.13	61.16	62.96	64.94	67.92	70.52	72.22	73.52	75.52	77.07
84	54.02	56.95	56.95	62.02	63.83	65.83	68.82	71.44	73.15	74.46	76.47	78.04
85	54.81	57.77	57.77	62.88	64.7	66.71	69.73	72.36	74.08	75.4	77.43	79
86	55.61	58.59	58.59	63.73	65.57	67.6	70.63	73.28	75.02	76.35	78.38	79.97
87	56.41	59.41	59.41	64.59	66.45	68.48	71.54	74.21	75.95	77.29	79.34	80.93
88	57.21	60.23	60.23	65.45	67.32	69.37	72.45	75.13	76.89	78.23	80.3	81.9
89	58.02	61.06	61.06	66.32	68.19	70.26	73.35	76.06	77.82	79.18	81.25	82.86
90	58.82	61.88	61.88	67.18	69.07	71.15	74.26	76.98	78.76	80.12	82.21	83.83
91	59.62	62.71	62.71	68.04	69.94	72.04	75.17	77.91	79.69	81.06	83.16	84.79
92	60.43	63.54	63.54	68.9	70.82	72.92	76.08	78.83	80.63	82.01	84.12	85.76
93	61.23	64.36	64.36	69.77	71.7	73.81	76.99	79.76	81.57	82.95	85.08	86.73
94	62.04	65.19	65.19	70.63	72.57	74.71	77.9	80.69	82.5	83.9	86.03	87.69
95	62.85	66.02	66.02	71.5	73.45	75.6	78.81	81.61	83.44	84.84	86.99	88.66
96	63.66	66.85	66.85	72.36	74.33	76.49	79.72	82.54	84.38	85.79	87.95	89.62
97	64.47	67.69	69.22	73.23	75.21	77.38	80.63	83.47	85.32	86.74	88.91	90.59
98	65.28	68.52	70.06	74.1	76.09	78.27	81.54	84.39	86.26	87.68	89.87	91.56
99	66.09	69.35	70.9	74.97	76.97	79.17	82.46	85.32	87.2	88.63	90.82	92.53
100	66.91	70.19	71.75	75.84	77.85	80.06	83.37	86.25	88.13	89.58	91.78	93.49

Appendix C
Error Functions

The Gaussian probability density function is given by:

$$P(x) = \frac{1}{\sqrt{2\pi}} e^{\frac{-x^2}{2}}$$

Assuming, $\mu = 0, \sigma = 1$

$$Erf(x) = \int_{-\infty}^{x} \frac{1}{\sqrt{2\pi}} e^{\frac{-z^2}{2}} dz$$

Note that *Erf*(0) = 0.5, *Erf*(infinity) = 1
Complementary error function, *Erfc*, is given by:

$$Erfc(x) = \frac{1}{2} erfc\left(\frac{x}{\sqrt{2}}\right)$$

Note the lowercase *e*.
In statistics textbooks, the error function is typically defined as

$$erf(x) = \frac{2}{\sqrt{\pi}} \int_{0}^{x} e^{-z^2} dz$$

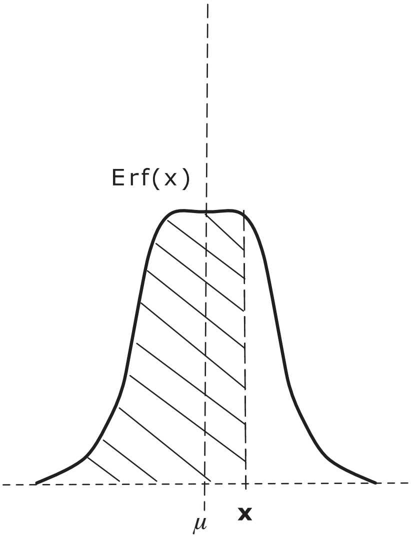

Figure C.1 Error function, *Erf*(*x*).

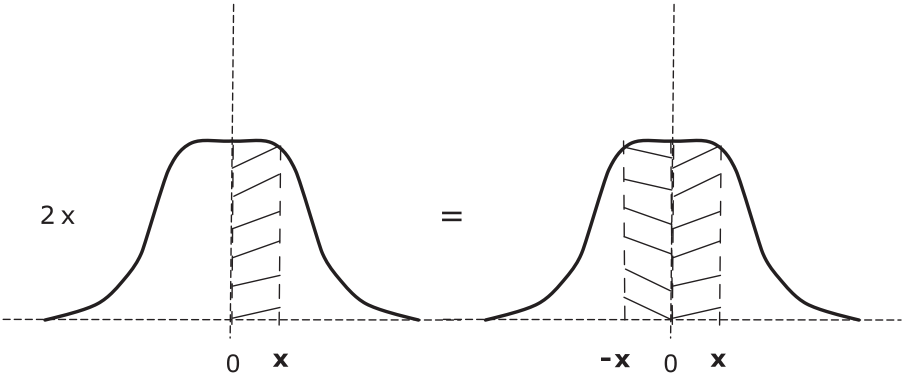

Figure C.2 Complementary error function, *Erfc*.

and

$$Erfc(x) = \frac{1}{2}\, erfc\left(\frac{x}{\sqrt{2}}\right)$$

$$erfc(x) = 2\, Erfc\left(\sqrt{2}x\right)$$

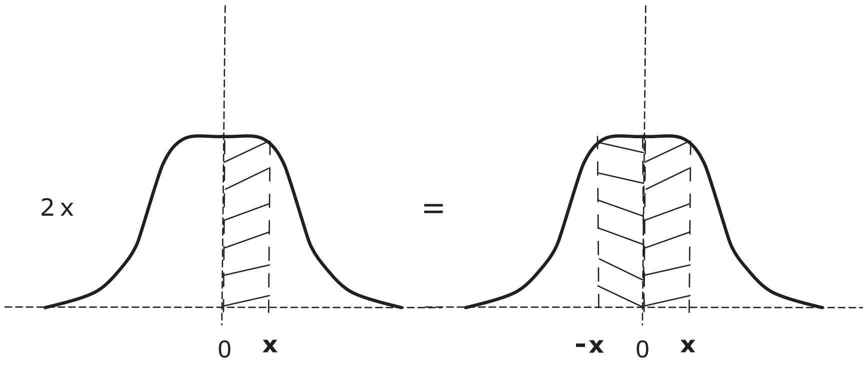

Figure C.3 *Erf* equivalents.

$$Q(x) = \int_{x}^{\infty} g(v)dv$$

$$erfc(x) = 2Q\left(x\sqrt{2}\right)$$

$$Q(x) = \frac{1}{2}\, erfc\left(\frac{x}{\sqrt{2}}\right)$$

$$erf\,(x) = \frac{2}{\sqrt{2}} \int_{0}^{x} e^{-z^2}\, dz = \frac{1}{\sqrt{\pi}} \int_{-x}^{x} e^{-z^2}\, dz$$

$$erfc(x) = 1 - erf\,(x) = \frac{2}{\sqrt{\pi}} \int_{x}^{\infty} e^{-z^2}\, dz$$

The Gaussian probability density function, when integrated between –*d* and *d*, for example, is shown in Figure C.5.

The probability that the Gaussian random function *x* takes values between –*d* and *d* is given in terms of the *erf* function as follows:

$$erf\left(\frac{d - \mu}{\sqrt{2}\sigma}\right)$$

Hence, the Gaussian error integral or *Q* function is:

$$Q(x) = \int_{x}^{\infty} g(v)dv$$

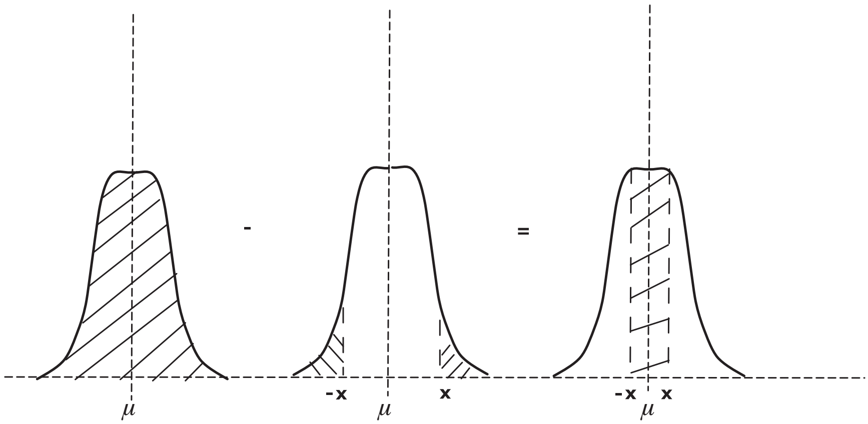

Figure C.4 *Erfc* equivalents.

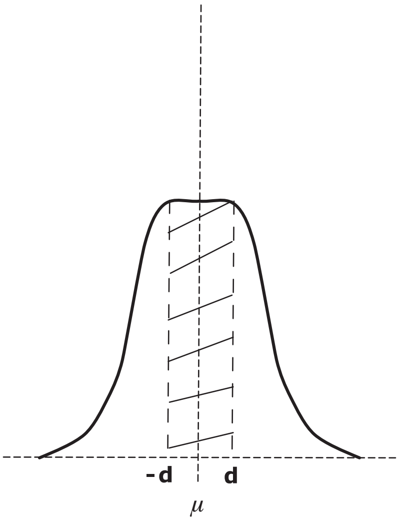

Figure C.5 Probability between random variable –*d* and *d.*

$$erfc(x) = \frac{2}{\sqrt{\pi}} \int_{x}^{\infty} e^{-v^2}\, dv$$

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_{x}^{\infty} e^{\frac{-v^2}{2}}\, dv$$

Therefore:

$$erfc(x) = 2Q\left(x\sqrt{2}\right)$$

$$Q(x) = \frac{1}{2}\,erfc\left(\frac{x}{\sqrt{2}}\right)$$

The error and complementary error functions are given in Tables C.1 and C.2.

Table C.1

Error Function, *Erf* = 1 – 0.5·*erfc*((*x*)/sqrt(2))

x	0.00	0.01	0.02	0.03	0.04	0.05	0.06	0.07	0.08	0.09
0.0	0.50000	0.50399	0.50798	0.51197	0.51595	0.51994	0.52392	0.52790	0.53188	0.53586
0.1	0.53983	0.54380	0.54776	0.55172	0.55567	0.55962	0.56356	0.56749	0.57142	0.57535
0.2	0.57926	0.58317	0.58706	0.59095	0.59483	0.59871	0.60257	0.60642	0.61026	0.61409
0.3	0.61791	0.62172	0.62552	0.62930	0.63307	0.63683	0.64058	0.64431	0.64803	0.65173
0.4	0.65542	0.65910	0.66276	0.66640	0.67003	0.67364	0.67724	0.68082	0.68439	0.68793
0.5	0.69146	0.69497	0.69847	0.70194	0.70540	0.70884	0.71226	0.71566	0.71904	0.72240
0.6	0.72575	0.72907	0.73237	0.73565	0.73891	0.74215	0.74537	0.74857	0.75175	0.75490
0.7	0.75804	0.76115	0.76424	0.76730	0.77035	0.77337	0.77637	0.77935	0.78230	0.78524
0.8	0.78814	0.79103	0.79389	0.79673	0.79955	0.80234	0.80511	0.80785	0.81057	0.81327
0.9	0.81594	0.81859	0.82121	0.82381	0.82639	0.82894	0.83147	0.83398	0.83646	0.83891
1.0	0.84134	0.84375	0.84614	0.84849	0.85083	0.85314	0.85543	0.85769	0.85993	0.86214
1.1	0.86433	0.86650	0.86864	0.87076	0.87286	0.87493	0.87698	0.87900	0.88100	0.88298
1.2	0.88493	0.88686	0.88877	0.89065	0.89251	0.89435	0.89617	0.89796	0.89973	0.90147
1.3	0.90320	0.90490	0.90658	0.90824	0.90988	0.91149	0.91309	0.91466	0.91621	0.91774
1.4	0.91924	0.92073	0.92220	0.92364	0.92507	0.92647	0.92785	0.92922	0.93056	0.93189
1.5	0.93319	0.93448	0.93574	0.93699	0.93822	0.93943	0.94062	0.94179	0.94295	0.94408
1.6	0.94520	0.94630	0.94738	0.94845	0.94950	0.95053	0.95154	0.95254	0.95352	0.95449
1.7	0.95543	0.95637	0.95728	0.95818	0.95907	0.95994	0.96080	0.96164	0.96246	0.96327
1.8	0.96407	0.96485	0.96562	0.96638	0.96712	0.96784	0.96856	0.96926	0.96995	0.97062
1.9	0.97128	0.97193	0.97257	0.97320	0.97381	0.97441	0.97500	0.97558	0.97615	0.97670
2.0	0.97725	0.97778	0.97831	0.97882	0.97932	0.97982	0.98030	0.98077	0.98124	0.98169
2.1	0.98214	0.98257	0.98300	0.98341	0.98382	0.98422	0.98461	0.98500	0.98537	0.98574
2.2	0.98610	0.98645	0.98679	0.98713	0.98745	0.98778	0.98809	0.98840	0.98870	0.98899
2.3	0.98928	0.98956	0.98983	0.99010	0.99036	0.99061	0.99086	0.99111	0.99134	0.99158

Table C.1 (continued)

x	0.00	0.01	0.02	0.03	0.04	0.05	0.06	0.07	0.08	0.09
2.4	0.99180	0.99202	0.99224	0.99245	0.99266	0.99286	0.99305	0.99324	0.99343	0.99361
2.5	0.99379	0.99396	0.99413	0.99430	0.99446	0.99461	0.99477	0.99492	0.99506	0.99520
2.6	0.99534	0.99547	0.99560	0.99573	0.99585	0.99598	0.99609	0.99621	0.99632	0.99643
2.7	0.99653	0.99664	0.99674	0.99683	0.99693	0.99702	0.99711	0.99720	0.99728	0.99736
2.8	0.99744	0.99752	0.99760	0.99767	0.99774	0.99781	0.99788	0.99795	0.99801	0.99807
2.9	0.99813	0.99819	0.99825	0.99831	0.99836	0.99841	0.99846	0.99851	0.99856	0.99861
3.0	0.99865	0.99869	0.99874	0.99878	0.99882	0.99886	0.99889	0.99893	0.99896	0.99900
3.1	0.99903	0.99906	0.99910	0.99913	0.99916	0.99918	0.99921	0.99924	0.99926	0.99929
3.2	0.99931	0.99934	0.99936	0.99938	0.99940	0.99942	0.99944	0.99946	0.99948	0.99950
3.3	0.99952	0.99953	0.99955	0.99957	0.99958	0.99960	0.99961	0.99962	0.99964	0.99965
3.4	0.99966	0.99968	0.99969	0.99970	0.99971	0.99972	0.99973	0.99974	0.99975	0.99976
3.5	0.99977	0.99978	0.99978	0.99979	0.99980	0.99981	0.99981	0.99982	0.99983	0.99983
3.6	0.99984	0.99985	0.99985	0.99986	0.99986	0.99987	0.99987	0.99988	0.99988	0.99989
3.7	0.99989	0.99990	0.99990	0.99990	0.99991	0.99991	0.99992	0.99992	0.99992	0.99992
3.8	0.99993	0.99993	0.99993	0.99994	0.99994	0.99994	0.99994	0.99995	0.99995	0.99995
3.9	0.99995	0.99995	0.99996	0.99996	0.99996	0.99996	0.99996	0.99996	0.99997	0.99997
4.0	0.99997	0.99997	0.99997	0.99997	0.99997	0.99997	0.99998	0.99998	0.99998	0.99998

Table C.2

Complementary Error Function, *Erfc*, $Q(x) = 0.5 \cdot erfc((x)/\mathrm{sqrt}(2))$

x	0.00	0.01	0.02	0.03	0.04	0.05	0.06	0.07	0.08	0.09
0.0	0.50000	0.49601	0.49202	0.48803	0.48405	0.48006	0.47608	0.47210	0.46812	0.46414
0.1	0.48803	0.44433	0.44038	0.43644	0.43251	0.42858	0.42465	0.46017	0.46017	0.46017
0.2	0.48803	0.40517	0.40129	0.39743	0.39358	0.38974	0.38591	0.42074	0.42074	0.42074
0.3	0.48803	0.36693	0.36317	0.35942	0.35569	0.35197	0.34827	0.38209	0.38209	0.38209
0.4	0.48803	0.32997	0.32636	0.32276	0.31918	0.31561	0.31207	0.34458	0.34458	0.34458
0.5	0.29806	0.29460	0.29116	0.28774	0.28434	0.28096	0.27760	0.30854	0.30854	0.30854
0.6	0.26435	0.26109	0.25785	0.25463	0.25143	0.24825	0.24510	0.27425	0.27425	0.27425
0.7	0.23270	0.22965	0.22663	0.22363	0.22065	0.21770	0.21476	0.24196	0.24196	0.24196
0.8	0.20327	0.20045	0.19766	0.19489	0.19215	0.18943	0.18673	0.21186	0.21186	0.21186
0.9	0.17619	0.17361	0.17106	0.16853	0.16602	0.16354	0.16109	0.18406	0.18406	0.18406
1.0	0.15151	0.14917	0.14686	0.14457	0.14231	0.14007	0.13786	0.15866	0.15866	0.15866
1.1	0.12924	0.12714	0.12507	0.12302	0.12100	0.11900	0.11702	0.13567	0.13567	0.13567

Table C.2 (continued)

x	**0.00**	**0.01**	**0.02**	**0.03**	**0.04**	**0.05**	**0.06**	**0.07**	**0.08**	**0.09**
1.2	0.10935	0.10749	0.10565	0.10383	0.10204	0.10027	0.09853	0.11507	0.11507	0.11507
1.3	0.09176	0.09012	0.08851	0.08691	0.08534	0.08379	0.08226	0.09680	0.09680	0.09680
1.4	0.07636	0.07493	0.07353	0.07215	0.07078	0.06944	0.06811	0.08076	0.08076	0.08076
1.5	0.06301	0.06178	0.06057	0.05938	0.05821	0.05705	0.05592	0.06681	0.06681	0.06681
1.6	0.05155	0.05050	0.04947	0.04846	0.04746	0.04648	0.04551	0.05480	0.05480	0.05480
1.7	0.04182	0.04093	0.04006	0.03920	0.03836	0.03754	0.03673	0.04457	0.04457	0.04457
1.8	0.03362	0.03288	0.03216	0.03144	0.03074	0.03005	0.02938	0.03593	0.03593	0.03593
1.9	0.02680	0.02619	0.02559	0.02500	0.02442	0.02385	0.02330	0.02872	0.02872	0.02872
2.0	0.02118	0.02068	0.02018	0.01970	0.01923	0.01876	0.01831	0.02275	0.02275	0.02275
2.1	0.01659	0.01618	0.01578	0.01539	0.01500	0.01463	0.01426	0.01786	0.01786	0.01786
2.2	0.01287	0.01255	0.01222	0.01191	0.01160	0.01130	0.01101	0.01390	0.01390	0.01390
2.3	0.00990	0.00964	0.00939	0.00914	0.00889	0.00866	0.00842	0.01072	0.01072	0.01072
2.4	0.00755	0.00734	0.00714	0.00695	0.00676	0.00657	0.00639	0.00820	0.00820	0.00820
2.5	0.00570	0.00554	0.00539	0.00523	0.00508	0.00494	0.00480	0.00621	0.00621	0.00621
2.6	0.00427	0.00415	0.00402	0.00391	0.00379	0.00368	0.00357	0.00466	0.00466	0.00466
2.7	0.00317	0.00307	0.00298	0.00289	0.00280	0.00272	0.00264	0.00347	0.00347	0.00347
2.8	0.00233	0.00226	0.00219	0.00212	0.00205	0.00199	0.00193	0.00256	0.00256	0.00256
2.9	0.00169	0.00164	0.00159	0.00154	0.00149	0.00144	0.00139	0.00187	0.00187	0.00187
3.0	0.00122	0.00118	0.00114	0.00111	0.00107	0.00104	0.00100	0.00135	0.00135	0.00135
3.1	0.00087	0.00084	0.00082	0.00079	0.00076	0.00074	0.00071	0.00097	0.00097	0.00097
3.2	0.00062	0.00060	0.00058	0.00056	0.00054	0.00052	0.00050	0.00069	0.00069	0.00069
3.3	0.00043	0.00042	0.00040	0.00039	0.00038	0.00036	0.00035	0.00048	0.00048	0.00048
3.4	0.00030	0.00029	0.00028	0.00027	0.00026	0.00025	0.00024	0.00034	0.00034	0.00034
3.5	0.00021	0.00020	0.00019	0.00019	0.00018	0.00017	0.00017	0.00023	0.00023	0.00023
3.6	0.00014	0.00014	0.00013	0.00013	0.00012	0.00012	0.00011	0.00016	0.00016	0.00016
3.7	0.00010	0.00009	0.00009	0.00008	0.00008	0.00008	0.00008	0.00011	0.00011	0.00011
3.8	0.00006	0.00006	0.00006	0.00006	0.00005	0.00005	0.00005	0.00007	0.00007	0.00007
3.9	0.00004	0.00004	0.00004	0.00004	0.00004	0.00003	0.00003	0.00005	0.00005	0.00005
4.0	0.00003	0.00003	0.00003	0.00002	0.00002	0.00002	0.00002	0.00003	0.00003	0.00003

Appendix D
Mathematical Formula

D.1 Basic RF Design Formulas

Free Space Path Loss

$$PL_{fs} = 32.45 + 20\log_{10}(f_{MHz}) + 20\log(d_{Km})$$

where F_{MHz} is frequency in MHz and d_{km} distance in km.

Power Law Path Loss

$$L = 10n\log_{10}\left(\frac{d}{d_{ref}}\right) + L_{ref}$$

where d_{ref} is reference distance and L_{ref} is path loss at reference distance.

Total Noise and Thermal Noise

$$\begin{aligned} N_{total} &= N_{th} + BW \\ N_{th} &= kT \end{aligned}$$

where BW is the bandwidth, k is the Boltzmann's constant ($1.38 \cdot 10^{-23}$ J/K), and T is the temperature in Kelvin, K.

Noise Floor

$NoiseFloor = kTBNF$

where NF is the noise figure of the receiver.

Signal-to-Noise Ratio

$$\frac{S}{N} = \frac{E_b}{N_o} \cdot \frac{C}{BW}$$

where the ratio E_b/N_o is bit energy-to-noise spectral power density, BW is bandwidth, and C is carrier power.

Carrier-to-Noise Ratio

C/N = carrier level at receiver input/noise floor of the receiver

Interference Plus Noise (dB)

$$I + N$$

Note: Convert each component (I and N) to its decimal counterpart first, add them, and convert them back to dB.

Wavelength (m)

$$\lambda = \frac{c}{f}$$

where c is speed of light, 3×10^8 m/s; and f is frequency in Hz.

Isotropic Receiver Level, IRL (dBm)

$$IRL = S_{rx} - G_{rx-ant}$$

where S_{rx} is the receiver sensitivity, and G_{rx-ant} is the receive-antenna gain.

System Gain, G_{sys}

$$G_{sys} = EIRP - IRL$$

Maximum Allowable Path Loss, MAPL

$$MAPL = G_{sys} - M_{tot}$$

where M_{tot} is the total link margin.

Trigonometry Identity

$$\cos^2 \theta = \frac{1+\cos 2\theta}{2}$$

$$\cos^3 \theta = \cos\theta\left[\frac{1+\cos 2\theta}{2}\right] = \frac{1}{2}\cos\theta + \frac{1}{2}\cos\theta\cos 2\theta$$

$$\cos\theta\cos 2\theta = \frac{\cos(\theta-2\theta)+\cos(\theta+2\theta)}{2} = \frac{\cos\theta+\cos 3\theta}{2}$$

$$\cos^3 \theta = \frac{1}{2}\cos\theta + \frac{1}{4}[\cos\theta+\cos 3\theta] = \frac{3}{4}\cos\theta + \frac{1}{4}\cos 3\theta$$

$$\cos x\cos y = \frac{1}{2}\cos(x+y) + \frac{1}{2}\cos(x-y)$$

$$\sin x\sin y = -\frac{1}{2}\cos(x+y) + \frac{1}{2}\cos(x-y)$$

$$\sin x\cos y = \frac{1}{2}\sin(x+y) + \frac{1}{2}\sin(x-y)$$

$$\cos x\sin y = \frac{1}{2}\sin(x+y) - \frac{1}{2}\sin(x-y)$$

$$\sin(x \pm y) = \sin x\cos y \pm \cos x\sin y$$

$$\cos(x \pm y) = \cos x\cos y \pm \sin x\sin y$$

Acronyms

2.5G 2.5 generation

2G Second generation

3G Third generation

3GPP Global Partnership Project 3

3G-SGSN Third generation GPRS support node

4G Fourth generation

AAA Authorization authentication accounting

AAS Address allocation server

ACM Address complete message

AES Advanced encryption standard

AICH Acquisition indicator channel

AK Authentication key

AM Adaptive mode, amplitude modulation

AMC Adaptive modulation and coding

AOA Angle of arrival

AP Access point

ASK Amplitude shift keying

ASN Access service network

ASP Application service provider

ATM Asynchronous transfer mode

AuC Authentication center

AWGN Additive white Gaussian noise

BCH Broadcast channel

BCCH Broadcast control channel

BE Best effort

BER Bit-error rate

BS Base station

BSC Base station controller

BTS Base station transceiver

BW Bandwidth

BWA Broadband wireless access

CCCH Common control channel

CCMP Countermode/CBC-MAC Protocol

CD/CA-ICH Collision detection/channel assignment indication channel

CDMA Code division multiple access

CID Connection identifier

CN Core network

CPCH Uplink common packet channel

CPE Customer premise equipment

CPICH Common pilot channel

CTCH Common traffic channel

CSICH CPCH status indication channel

CSN Connectivity service network

DBPSK Differential binary phase-shift keying

DCCH Dedicated control channel

DCH Dedicated transport channel

DES Data encryption standard

DHCP Dynamic Host Configuration Protocol

DL Data link

DOA Direction of arrival

DOS Denial of service

DPCCH Dedicated physical control channel

DPDCH Dedicated physical data channel

DQPSK Differential quadrature phase-shift keying

DSCH Downlink shared channel

DSCP Differentiated service code point (also DiffServ)

DSL Digital subscriber line

DSP Digital signal processing

DSSS Direct sequence spread spectrum

DTCH Dedicated traffic channel

DU Dense urban

EAP Extensible Authentication Protocol

ECSD Enhanced circuit switch data

EDGE Enhanced data rates for GSM evolution

EGPRS Enhanced GPRS

EIRP Effective isotropic radiated power

EPC Evolved packet core

ErtPS Extended real-time polling service

ERT-VR Extended real-time variable rate

ETH-CS Ethernet convergence sublayer

E-UTRAN Evolved-UTRAN

EVDO Evolution data option

FACH Forward access channel

FDD Frequency division duplex

FDMA Frequency division multiple access

FEC Forward equivalency class; forward error correction

FM Fade margin; frequency modulation

FSK Frequency shift keying

FTP File Transfer Protocol

FUSC Full usage of subchannel

GERAN GSM/EDGE radio access network

GHz Gigahertz

GOS Grade of service

GPRS General packet radio service

GRE Generic routing encapsulation

GSM Global system for mobile communication

GTT Global title translation

HLR Home location register

HMAC Hash message authentication code

HSCSD High-speed circuit-switched data

HSDPA High-speed downlink packet access

HSPA High-speed packet access

HSUPA High-speed uplink packet access

HTTP Hypertext Transfer Protocol

IAM Initial address message

ICI Intercarrier interference

IETF Internet Engineering Task Force

IF Intermediate frequency

IFFT Inverse fast Fourier transform

IM Intermodulation

IMS IP multimedia subsystems

IN Intelligent network

IntServ Integrated service

IP-CS IP-convergence sublayer

IRL Isotropic receive level

IS-95 Interim standard

ISDN Integrated service digital network

ISI Intersymbol interference

ISUP ISDN user part

ITU International Telecommunication Union

IWF Interworking function

JPEG Joint Photographic Experts Group

kbps Kilobits per second

KEK Key exchange key

LAPM Link access procedure for modems

LMDS Local multipoint distribution systems

LOS Line of sight

LTE Long-term evolution

MAC Media access control

MAPL Maximum allowable path loss

Mbps Megabits per second

MGCP/MEGACO Media Gateway Control Protocol/Media gateway controller

MIM Man in the middle

MIMO Multiple input multiple output

MMDS Multichannel-multipoint distribution systems

MPEG Motion Picture Experts Group; Moving Picture Experts Group

MPLS Multiprotocol Label Switching

MRC Maximal ratio combiner

MS Mobile station

MSC Mobile switching center

MSC/VLR Mobile switching center/visitor location register

MTP Message transfer part

NAP Network access provider

NAS Network access server

NF Noise figure

NLOS Nonline of sight

NRM Network reference model

nrtPS Nonreal-time polling service

NSP Network service provider

NW Network

OFDM Orthogonal frequency division multiplexing

OFDMA-PHY Orthogonal frequency-domain multiple-access physical layer

OMAP Operation, maintenance, and administration part

OSI Open system interconnection

OVSF Orthogonal variable spreading factor

PCCH Paging control channel

PCCPCH Primary common control physical channel

PCH Paging channel

PCM Pulse code mode modulation

PCPCH Physical common packet channel

PDSCH Physical downlink shared channel

PDU Packet data unit

PICH Paging indication channel

PK Public key

PKI Public-key interchange

PKMv2 Private key management

PLMN Public land mobile network

PMK Pairwise master key

PN Pseudonoise

POP3 Post Office Protocol version 3

POT Plain old telephone

PPP Point-to-Point Protocol

PRACH Physical random access channel

PSK Phase-shift keying

PSTN Public switched telephone network

PtMP Point to multipoint

PTP Point to point

PUSC Partial usage of subcarriers

QAM Quadrature amplitude modulation

QoS Quality of service

R Rural

RACH Random-access channel

RADIUS Remote authentication dial-in user service

RF Radio frequency

RNC Radio network controller

RRC Radio resource management

RSA Ron Rivest, Adi Shamir, and Leonard Adleman

RSN Robust security network

RTP Real-Time Protocol

rtPS Real-time polling service

SA Security association

SCCP Signal connection control part

SCCPCH Secondary common control physical channel

SUIRG Satellite user interference reduction group

S-OFDMA Scalable orthogonal frequency domain multiple access

SCH Synchronization channel

SCP Service control point

SDS Significantly degraded second

SDU Service data unit

SFID Service flow identifier

S-GW Serving gateway

SHCCH Shared channel control channel

SINR Interference and noise ratio

SIP Session Initiation Protocol

SM Spatial multiplexing

SMC Signal modulation scheme

SMTP Simple mail transfer protocol

SNR Signal-to-noise ratio

SS7 Signaling system 7

SSP Service switching point

STBC Space-time block coding

STP Signal transfer point

SUI Stanford University Interim

SIM Subscriber identity module

SS Subscriber station

SU Suburban

TCAP Transaction-capability application

TCP Transport Control Protocol

TDD Time-division duplex

TDMA Time-division multiple access

TKIP Temporal Key Integrity Protocol

TMAs Tower-mounted amplifiers

TP Transport

TUP Telephone user part

U Urban

UDP User Datagram Protocol

UE User equipment

UGS Unsolicited grant services

UMB Ultramobile broadband

UMTS Universal mobile telecommunication system

UNII Unlicensed national information infrastructure

UTRAN UMTS terrestrial radio access network

VAS Value-added services

VoIP Voice-over-Internet Protocol

WCDMA Wideband CDMA

WEP Wireless equivalent privacy

WiBro Wireless broadband

WiFi Wireless fidelity

WiMAX Worldwide interoperability microwave access

WISP Wireless Internet service provider

WLAN Wireless local area network

WLL Wireless local loop

WMANs Wireless area metropolitan networks

WPA WiFi protected access

About the Author

Zerihun Abate is a wireless communications engineer with more than 14 years of experience in wireless systems engineering and RF network planning. He has a B.S.E.E. and an M.S.E.E. from George Mason University. Mr. Abate has worked for big technology players such as Lucent, Siemens, and Qualcomm, and he is currently a wireless communications engineering consultant.

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