A PROPOSED ARCHITECTURE FOR A HIGH-DATA RATE MOBILE LMDS NETWORK

By
Katina Roshael Reece

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Charles Bostian, Chair
Nathaniel Davis, IV
Timothy Pratt
Jeffrey Reed

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(ABSTRACT)

This thesis proposes a system architecture for a high-data rate mobile Local Multipoint Distribution Service (LMDS) Network. Its goal is to provide a workable "strawman" design that can serve as a basis for further research. The architecture is designed to offer broadband services to train commuters via LMDS. The thesis provides a broad overview of system aspects, such as Doppler shift, modulation selection, and error correction coding. These parameters and others are important in providing a robust design.

This thesis discusses a set of criteria that seek the best approach in terms of economical feasibility, throughput capabilities, design complexity, data routing, and robustness in serving multiple mobile units simultaneously. These criteria are examined through link budget analysis, layout designs, and throughput efficiency.

System throughput specifications are calculated for services, such as web browsing, email, ftp, and voice services to 100 train commuters. The information rate was 134.4 Mbps. The information rate plus overhead, which includes routing, bit and framing synchronization, and error correction coding, was approximately 201.51 Mbps. Using Carson's rule, the total required bandwidth for downstream transmissions was approximately 263 MHz. This throughput requirement was a criterion in selecting the appropriate system architecture.

Three approaches were evaluated: LMDS Infostations, Tower Sites, and Infostations/Tower Sites. Infostations are low-powered wireless cells designed to offer individual pockets of high bandwidth connectivity for broadband services. Tower Sites use switched antenna beams to offer continuous services to train commuters. The hybrid solution, Infostations/Tower Sites, offers

continuous services with increased power requirements and increased base stations separation when compared to the Infostation approach.

Link budgets were examined for the Infostations and Tower Site approach. The initial required power for the Infostation was 1 mW. A 42.4 dB E_b/N_o link margin was computed using the Friis equation. The initial required power for the Tower Site approach was 500 mW. A 10.2 dB E_b/N_o link margin was computed with this approach. Tradeoffs with the non-fixed parameters were made to vary the link margins.

An economically feasible number of required units were also determined. Approximately 3,000 low-powered Infostations would be needed to offer continuous service. Only 93 Tower Sites would be required and 4*93 Tower Site/Infostation units would be needed to supply continuous, seamless services over a 230-miles coverage area.

The LMDS Tower Site was chosen to be the most suitable approach because of its robustness in meeting the pre-defined criteria.

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Chapter 1

Introduction

1.1 Motivation

Today, more and more consumers desire ubiquitous services, such as web browsing and electronic mail (e-mail) services. The world of telecommunications is realizing these desires through research, testing, and development. The advent of universal services via satellites on airplanes and other mobile units is just the tip of future technological advancements. Why not offer users the capability to search the web, download images, send and retrieve email messages "on the fly," and use typical telephony services in yet another environment, trains, via another medium, Local Multipoint Distribution Services (LMDS)? What better way to utilize the large LMDS bandwidth than to offer consumers the ability to use common LAN-based equipment aboard trains and in other environments?

1.2 Problem Definition

LMDS was designed to offer "fixed" high data rate services to homes and businesses in the last mile, where bandwidth capacity bottlenecks are most common. This thesis focuses on modulation selection, Doppler shift, error correction coding, communication establishment, data routing, and other issues relevant to a "mobile" environment. It investigates these issues from a broad perspective and offers recommendations for a realistic prototype.

1.3 Summary of Approach

The goal of this thesis is to provide an overview of design issues and approaches for a high-speed mobile, LMDS network. Envision boarding the LMDS-MARC train in Blacksburg, VA, and realizing that you neglected to include an important diagram in the PowerPoint presentation you will give in a half hour. This presentation could lead to the acquisition of a substantial amount of funds from a contractor, "Reece R Us," for the next five years. The diagram is located on a server in your local home directory at Virginia Tech on "lmac.ee.vt.edu." Due to other technicalities, it is necessary for you to use the telephone services and your file transfer program to acquire the file. Using an on-board Ethernet LAN port located in the seatback on the commuter train, you are able to download the presentation successfully.

This scenario could become a typical daily event. Figure 1.1 offers an end-to-end glimpse of this overall enabling system architecture. The Internet services will be provided via an Internet Service Provider (ISP) or the Internet itself. This data will next be forwarded in Asynchronous Transfer Mode (ATM) packages from an ATM network to an ATM switch. We will assume that the ATM units will have mobility protocol capabilities that ensure the flow of data among LMDS Infostation Radio Ports (LIRPs). The ATM packages will be sent via wire-line connections to an LMDS Infostation controller. This controller then forwards all packages to the designated LIRPs. Next, the data is transmitted from the ATM switch over the LMDS frequency spectrum to the mobile transceiver(s) on the train using a pre-determined data package format and duplexing technique. If mobile ATM is used, then a standard ATM user-network interface (UNI) will be connected to the train terminals as well. The final transmission is completed via an Ethernet hub, which then forwards the data packet(s) to the appropriate user's station.

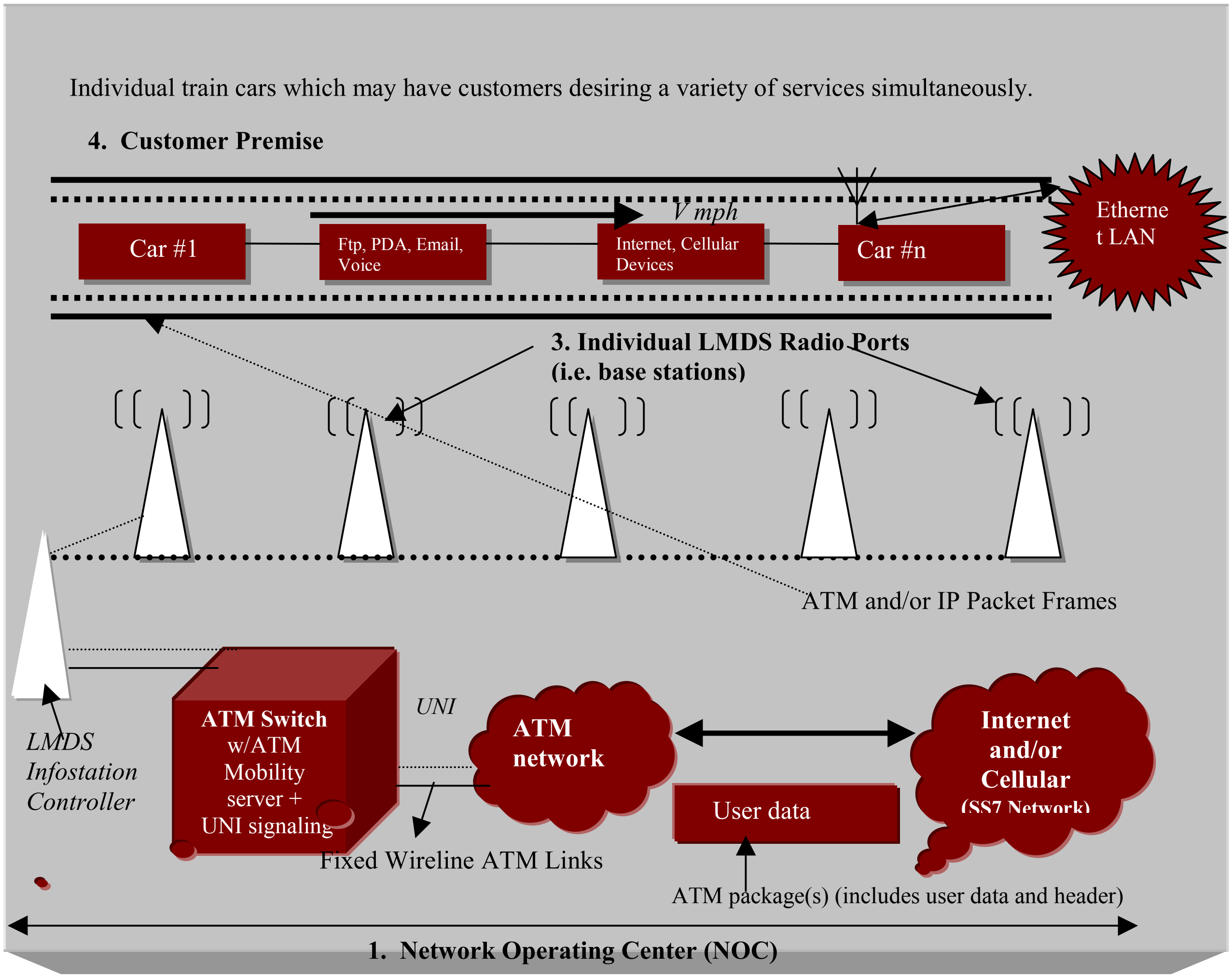

Figure 1.1: An End-To-End LMDS, Mobile System Architecture using ATM

The description above offers a good model for the system architecture. This thesis focuses on the internetworking between areas three and four and the data transmissions in the LMDS channels. Issues that were considered as salient problems in our high-speed mobile LMDS channel are listed below.

- **Data Rate** – What is the required data rate for each user? What is the cumulative data rate?
- **Modulation Selection** – What is the most suitable modulation technique? What is the appropriate error correction technique?

- **Overhead Requirements –** What overhead is encountered, including cumulative data rate, data routing, timing synchronization, and error correction selection?
- **Channel Issues** – How will propagation conditions and Doppler effect the channel?
- **Bandwidth** – What is the required bandwidth considering the previously mentioned issues?
- **Link Design** – What are the appropriate parameters in obtaining a pre-defined C/N and E_b/N_o?
- **Hardware Issues** – What power levels, antenna gains, and noise temperatures are required?

The composition of these issues provides a basis for the actual implementation of a high-speed mobile LMDS channel.

1.4 Contribution of Research

This thesis provides three potential solutions for a system architecture in deploying a mobile LMDS network. It gives recommendations for key physical layer components (i.e. LMDS channels). The LMDS Infostation approach proves to be a feasible solution, but not the most economical. The LMDS Tower Site approach proves to be the most robust in meeting a set of pre-defined criteria. Finally, this thesis provides insight for future standards and deployment of mobile high-speed LMDS architectures.

1.5 Thesis Overview

Chapter 2 discusses physical layer background alternatives associated with implementing LMDS in a mobile environment. Each section presents a preferred selection applicable to our system architecture. Chapter 3 reviews a set of system specifications for our various methodologies. Chapter 4 discusses three system design approaches: LMDS Infostations, LMDS Tower Sites, and LMDS Infostation/Tower Site. Networking issues are also discussed in this chapter. Finally, Chapter 5 offers recommendations for further research. The goal of this methodology is to stimulate an awareness of issues relative to the actual deployment of LMDS in a mobile high-speed environment.

Chapter 2

Background

LMDS is a wireless, asymmetric broadband technology in the 28 – 31GHz frequency range. See Figure 2.1 for a specific layout of the frequency spectrum as allocated by the Federal Communication Commission (FCC). [3] With the available 1.3GHz bandwidth, transmission data rates can exceed 1Gbps.

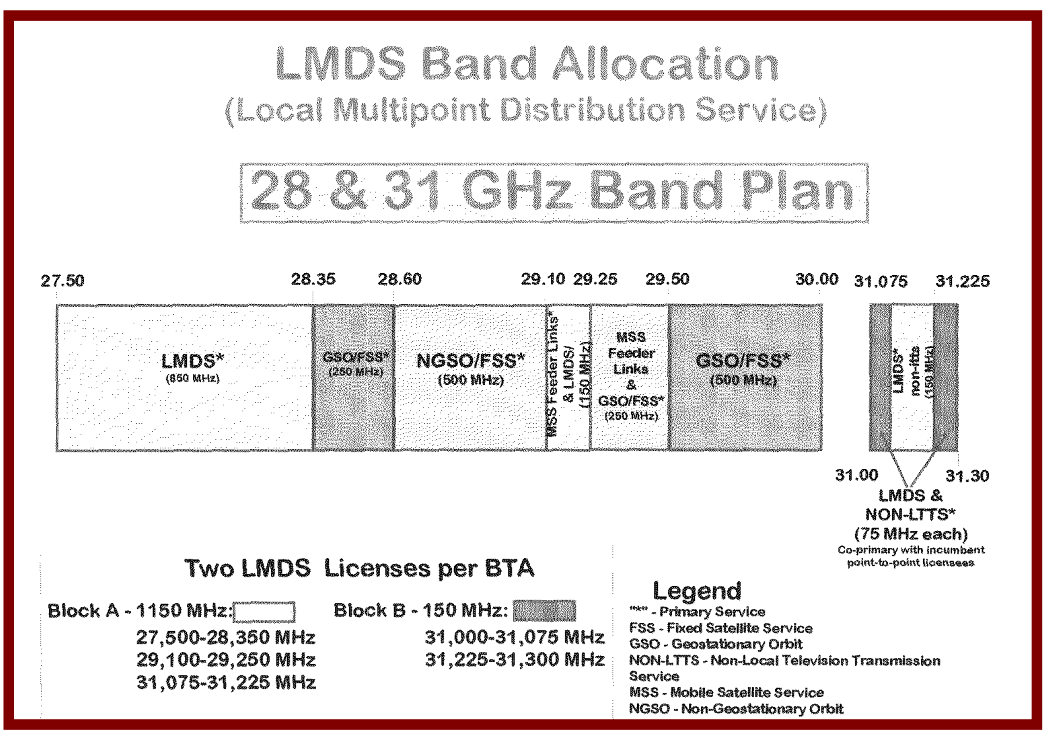

Figure 2.1: LMDS Spectrum Allocation [3]

The propagation characteristics associated with this frequency range force the transmitter and receiver to be within **L**ocal proximity of each other. Since the signal can be broadcast to multiple users simultaneously, it is called **M**ultipoint. The applications include a **D**istribution of **S**ervices, such as video, telephony, and Internet traffic.

The basic components of a fixed LMDS network typically include four major parts: Network Operations Center (NOC), fiber-based infrastructure, base station, and customer premise equipment. The NOC controls the network management functions of customers, such as the Asynchronous Transfer Mode (ATM) and Internet Protocol (IP) switching and any required interconnection to the Internet and the Public Switch Telephone Network (PSTN). The fiber based infrastructure consists of Synchronous Optic NETwork (SONET) speed rates, such as OC-12 and OC-3 links. The conversion from the fiber infrastructure to the wireless network occurs at the LMDS base stations. Finally, the customer equipment commonly consists of outdoor mounted microwave equipment and any required indoor digital equipment. This indoor digital equipment typically consists of modulation, demodulation, control, and any customer required interface. [1]

Because our primary focus is using "mobile" LMDS, we consider three main network entities similar to the terrestrial LMDS network. The three main networks in this design are as follows: 1) the backbone terrestrial network connecting the wired infrastructure to the LMDS base stations, 2) the LMDS network connecting the LMDS base stations to the transceivers on the moving unit, and 3) the internal network of the mobile unit (i.e. train). The thesis is concerned primarily with issues relative to the second network, with a particular focus on the physical layer.

2.1 Physical Layer Features

It is important to understand the physical layer issues that affect the seamless flow of data in an LMDS network, such as propagation factors, modulation, and channel coding. The project first focuses on channel frequency issues, including Doppler effect and propagation conditions.

2.1.1 Channel Characteristics

2.1.1.1 Doppler shift

Doppler shift is the change in frequency proportional to the speed and direction of the moving object. [4] It is not a new concept in wireless communications, satellite systems often undergo Doppler shift as a satellite moves relative to its target receiver (s). Doppler shift will be positive when the mobile is moving towards the LMDS base station (i.e. the perceived frequency is increased) and negative when the mobile is moving away from the LMDS base station. Below is an example of the potential change in frequency that could occur between the train and the LMDS radio ports.

If we assume a carrier frequency, f_c, of 27.5GHz, then this results in a wavelength of $\lambda=(c/f_c)$ = 3e8 /27.5e9 = 11mm. Table 2.1 gives the sample calculations.

Table 2.1: Doppler Effects of Received Frequency at LMDS Spectrum

Scenarios	**Velocity(v)**	Received Frequency (mobile travels towards transceiver), Doppler shift (fd) is positive **$f = f_c+f_d$, where $f_d = v/\lambda$**	Received Frequency (mobile travels away from transceiver), Doppler shift is negative **$f = f_c- f_d$**	Doppler shift as a percentage of **fc. (fc/fd)*100**
Worst Case	200 mph =89.4 m/s	f=27.5e9 + 8195 Hz =27.5000082GHz	27.4999918GHz	2.98e-05%
Typical Case	120 mph= 53.66 m/s	f= 27.5e9 + 4878 Hz = 27.5000048GHz	27.4999951GHz	1.77e-05%
Minimum Case	80 mph = 35.77 m/s	f = 27.5e9+3252 Hz = 27.5000032GHz	27.4999967GHz	1.18e-05%

In addition to the Doppler effect, we will also have to consider channel attenuation.

2.1.1.2 Attenuation

At LMDS frequencies, most fading is due to rain attenuation, vegetation attenuation, and scattering; however, the effect of rain will be most important in our network. Rain causes a reduction in the received signal level because individual raindrops absorb energy from radio waves. One method to reduce the effects of rain fade is to design the links with sufficient link margin or excess power to overcome rain fades. Other approaches involve an adaptive solution whereby a feedback system automatically increases the transmitting power when fading occurs. This method allows the transmitted power to be limited and held in reserve until the link needs it. LMDS signals may better withstand rain fade because of the accommodation in power adjustment and architectural layout.

We will now discuss the procedural steps to calculate rain attenuation.

Step 1: Calculate the *specific attenuation., A.,* Eq. 2.1

$$A_{dB/km} = aR^{b} \tag{2.1}$$

where,

a and *b* depend strongly on frequency. An approximation for these values can be found in [7].

R is the rain rate measured in (mm/hr). We must select the rain rate for the appropriate exceedance and geographical region. Exceedance is the performance metric of the radio link relative to rain attenuation. For example, an exceedance of 0.01% implies that the link will be available for 99.99 % of the time and unavailable for 0.01% of the time.

Step 2: Determine the path averaging factor, Eq. 2.2.

$$r = \frac{1}{1 + \frac{d}{d_o}} \tag{2.2}$$

where,

d = path length of the radio link or the right-of-way distance of the train.

d_o is determined by Eqs. 2.3 – 2.4.

$$d_o = 35e^{-0.015R_{0.01}};\ R_{0.01} \le 100mm/hr \tag{2.3}$$

$$d_o = 35e^{-0.015x100};\ R_{0.01} \ge 100mm/hr \tag{2.4}$$

Step 3: Compute the attenuation with a certain exceedance (ex. 0.01 %), Eq. 2.5.

$$A_{0.01} = A_{dB/km} \times d \times r \tag{2.5}$$

where the variables in Eq. 2.5 were determined from Eqs. 2.1 – 2.4.

Step 4: Calculate the rain attenuation for other exceedance values, Eq. 2.6.

$$A_p / A_{0.01} = 0.12 \times p^{-(0.546+0.043 \log 10 p)} \qquad (2.6)$$

where,

A_p is the attenuation at exceedance p.

As indicated earlier, rain attenuation is a problem at LMDS frequencies. Thus, we must consider the effects of rain attenuation in our train environment. Because of the close proximity of the transmitter and the receiver, the attenuation can be overcome by increasing the power usage during excessive rain periods. Chapter 4 discusses various link budgets, which reveal large link margins to compensate for rain attenuation.

Another important characteristic of the physical layer is modulation because it is this process that actually carries the information in the LMDS channel.

2.1.2 Modulation

Modulation is a technique whereby digital information is sent over a desired transmission medium. This process may be implemented by varying the amplitude, phase, or frequency of a high frequency carrier in accordance with the information in the message signal. [4] The choice of modulation depends on several factors, such as the desired Bit Error Rate (BER), channel conditions, propagation issues, and available bandwidth. Thus, the ability to deliver error-free transmission is heavily influenced by the chosen modulation. A fundamental tradeoff in digital communication systems between power efficiency and bandwidth efficiency is evident in the modulation choice. We briefly discuss this tradeoff before evaluating the modulation selections.

2.1.2.1 Power Efficiency and Bandwidth Efficiency

One of the factors in modulation selection is influenced by the tradeoff between bandwidth efficiency and power efficiency. Bandwidth efficiency is measured by the number of bits per second transmitted per hertz of bandwidth. Fortunately, with the use of LMDS, bandwidth will not be a major limiting factor, but we still wish to use the available bandwidth efficiently. LMDS has approximately 1.3GHz of bandwidth available for asymmetric and/or symmetric transmissions. This bandwidth can serve high data rate multimedia applications to a large number of end users simultaneously.

Power efficiency describes the ability of the selected modulation to maintain a desired signal fidelity at low power levels. In other words, in order to obtain a good quality signal, one must use a certain level of signal power while maintaining an acceptable Bit Error Rate (BER). This tradeoff is often measured as a function of energy per bit vs. noise power density (E_b/N_o) required at the receiver's input.

2.1.2.2 Evaluation of Modulation Selection

The characteristics of the modulation technique will determine the performance of our system architecture in the depicted environment. The ideal case would be to test various modulations in a mobile high-speed LMDS channel. The main categories for modulation schemes are those that require coherent detection, such as phase shift keying (PSK), and those that require non-coherent detection, such as frequency shift keying (FSK).

2.1.2.2.1 Non-Coherent Frequency Shift Keying (N-FSK) Demodulation Process

In past years, non-coherent FSK (N-FSK) has been a popular choice for signals in wireless environments where the receiver is moving at various high speeds (ex. 120mph – 160mph) relative to the transmitter. This is because the only synchronization requirements are bit timing and frequency tracking (see Section 2.1.2.2.2 on synchronization issues) with N-FSK. Figure 2.2 gives a general idea of the required components for non-coherent FSK detection. Bandpass filters are used to extract only the signal in the effective bandwidth, B_p. An envelope detector is used as an energy detector for the two frequency tones (i.e. binary case). FSK is typically implemented as orthogonal signaling whereby each frequency tone in the signal set does not interfere with any of the other tones. [12] The final output stage is implemented using an optimum threshold algorithm to obtain the original signal. Because of the complexity involved in demodulating at high frequencies, we need to have intermediate frequencies (IF) to bring the signal to baseband. Typical final stage IF values for LMDS frequencies are between 2.0 and 3.0GHz [17].

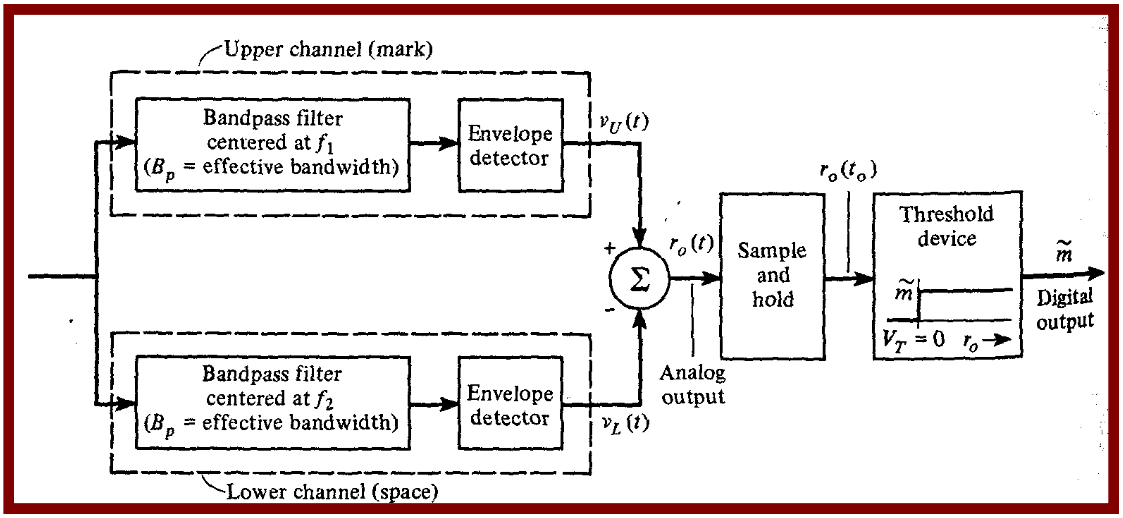

Figure 2.2: Non-Coherent Receiver Detection of FSK. [15]

2.1.2.2.2 Timing Synchronization

All digital communication systems involve three levels of synchronization: bit, frame or word, and carrier synchronization. These synchronization signals are clock-type signals that are delayed when compared to the clock signals at the transmitter due to the propagation delay through the channel. Both bit and frame synchronization are required in all modulation schemes, but carrier synchronization is necessary only with coherent detection. Thus, in PSK modulation the biggest problem is in the ability of the demodulators phase locked loop to synchronize with the transmitted signals carrier reference. Until synchronization is gained, bits will be lost, decreasing the throughput of the system.

Bit synchronization refers to the receiver's ability to distinguish one bit interval from another. Bit synchronizers are used to provide a bit sync signal for clocking the sample and hold circuit. In N-FSK, the bit sync signal can be used for clocking the matched filter circuit. [15] The matched filters are modeled as bandpass filters centered at f_1 and f_2 (see Figure 2.2). The sampling occurs at the output of the envelope detectors, which are sampled at every $t=kT_b$, where k is an integer and T_b is the bit period. [4] Thus, the decision of which M–ary N-FSK signal was transmitted depends on which envelope detector has the maximum output.

Finally, the frame and word synchronization are needed because we have proposed to use ATM (i.e. frame synchronization) and FEC (i.e. word synchronization). This process will occur at baseband. Frame synchronization is needed because ATM takes advantage of dynamic TDM and it uses fixed sized cells. Multiple ATM cells may be packetized into one frame thus creating a need to distinguish each frame. The FEC systems using block or convolutional coding requires a clock in the receiving decoding circuits for word synchronization. [15] These synchronization factors will induce an overhead processing delay and will be evaluated in Chapter 3.

Having discussed typical synchronization issues, we will next examine the advantageous of using FSK in handling the effects of Doppler.

2.1.2.2.3 N-FSK and Doppler Dependency

Non-coherent FSK handles the Doppler effects more efficiently than PSK because of the differences in synchronization demands. N-FSK only has to track the carrier frequency. PSK has to track the incoming phase. Essentially, the Doppler-shifted signal must remain within the bandwidth of the matched filter. This is especially difficult when a signal is undergoing large Doppler shifts or suffering deep fades. This could result in a more complex design problem for the LMDS mobile transceivers. [12] Non-coherent FSK is also more power efficient than PSK. It is true that PSK is more bandwidth efficient. However, the power usage is more of an issue than bandwidth conservation in our system architecture.

We will next use the probability of error formulas for terrestrial digital communication systems to compare coherent and non-coherent modulation techniques. Table 2.2 gives the relationship between probability of error, P_e and E_b/N_o for both coherent PSK and N-FSK. Table 2.3 gives the relationship between bandwidth and transmission bit rate for these modulations.

Table 2.2: P_e vs. E_b/N_o for Respective Modulation Types

Modulation	**Probability of Error, P_e**
Coherent PSK	$Q\left(\sqrt{\frac{2E_b}{No}}\right)$ (2.7)
M-ary Noncoherent FSK	$P_e \leq \frac{M-1}{2}\exp\left(-\frac{E_s}{2N_o}\right)$, $E_s = kE_b$, $M=2^k$ [4] (2.8) where, k = number of bits/modulation symbol, M = number of symbols, E_s = amount of energy per symbol, E_b = amount of energy per bit, N_o = noise spectral density.

Table 2.3: Bandwidth Efficiency for Respective Modulation Types at BandPass

Modulation	**Bandwidth Efficiency R_b/B (b/sHz)**
Coherent PSK	B = $\frac{R_b(1+r)}{\log_2 M} + 2f_d$ (2.9) where, R_b = transmission rate in bits/second, r = raise cosine factor, f_d = Doppler frequency shift and depends on the speed of the mobile unit and the wavelength (see Table 2.1.)
Non-coherent FSK	B = $\frac{RbM}{2 * \log_2 M}(1+r) + 2f_d$ (2.10)

Table 2.4 and Table 2.5 show the tradeoff in bandwidth and power efficiency for both PSK and FSK using the above formulas.

Table 2.4: Bandwidth and Power Efficiency of M-ary PSK Signals [4]

M	2	4	8	16	32	64
$\eta_B = R_b/B$*	1	2	3	4	5	6
E_b/N_o for BER =$10e^{-6}$	10.5	10.5	14	18.5	23.4	28.5

*B represents the first null-bandwidth of M-ary PSK signals

Table 2.5 Bandwidth and Power Efficiency of M-ary FSK Signals

M	2	4	8	16	32	64
$\eta_B = R_b/B$	1	1	.75	.5	.33	.19
E_s/N_o for BER = $10e^{-6}$	13.35	13.77	14.07	14.3	14.55	14.76
E_b/N_o for BER = $10e^{-6}$	13.35	10.76	9.3	8.3	7.55	7

Table 2.4 and Table 2.5 show that M-ARY N-FSK is more power efficient because as M increases, less power is required at the receiver per bit. This is because FSK signal space is less crowded than PSK and thus does not require a lot of power to distinguish each symbol. That is, FSK signals are more orthogonal which implies that these signals are “independent” of each other.

The choice of N-FSK is thus supported through calculations and theoretical concepts. However, selecting the most suitable modulation type is not the only mechanism of reducing the effects of channel conditions in a digital communication system. The next section will briefly discuss common error correction methods.

2.1.3 The Process of Error Correction

2.1.3.1 Interleaving

The process of interleaving the coded message before transmission spreads bursts of channel errors out in time. This enables the decoder to handle them as if they were random errors without adding any overhead. Interleaving protects important source data during deep fades or noise bursts; it spreads the bits out over time to prevent groups of important data from being lost. The interleaver scrambles the time order of source bits before channel coding since the channel codes are designed to protect against errors that may occur in a bursty manner. [12]

2.1.3.2 Channel Coding

Channel coding improves communications performance by enabling the transmitted signals to better withstand the effects of channel impairments. Typically, the ultimate goal of channel coding is to reduce the P_e or the required E_b/N_o at the cost of expending more bandwidth than is otherwise required. Channel coding is implemented by selectively introducing redundant bits in the transmitted data. Error detection codes are channel codes that detect errors. Error correction codes can both detect and correct errors. [4] There are many different types of error codes, such as Reed-Solomon (RS). They are commonly used to correct errors that normally occur in bursts.

We will assume the use of a concatenated code based on a convolution code and a shortened RS code, as was suggested by the, Digital Audio Visual Integrated Circuit (DAVIC). [10] Section 2.2.1.2 presents a more elaborate discussion on channel coding methods.

We have discussed the physical layer issues associated with the channel transmission model. We will next introduce the basic transmission theory and formulas used to evaluate the performance of communication systems.

2.1.4 Physical Transmitter and Receiver Link

2.1.4.1 Friis Equation

In designing our radio communication link, fundamental equations exist that aid in determining the system performance. First, an essential parameter in evaluating our total received power is our choice of antennas. Antenna gain is described by G(θ), where θ indicates the direction of transmitter. It is measured by the increase in power radiated by the transmitter over the amount of power transmitted using an isotropic antenna. An isotropic antenna radiates power equally in all directions. Equation 2.11 describes this relationship. [7]

$$G(\theta) = \frac{S(\theta)}{\frac{P_o}{4\pi}} \qquad (2.11)$$

where,

S (θ) = the power radiated per unit solid angle by the antenna in direction θ,

P_o = the total power transmitted by the test antenna,

G (θ) = the gain of the antenna angle θ.

We will use the Friis transmission equation, Eq. 2.12, to calculate the received power. This equation relates the received power, P_r, to the transmitted power, P_t, the transmitter gain, G_t, the receiver Gain, G_r (see Eq. 2.13), and the path loss, L_p. Equation 2.12 applies to propagation between ideal antennas in empty space.

$$P_r = P_t G_t G_r \left[\frac{\lambda}{4\pi R} \right]_2 \qquad (2.12)$$

The receiver gain depends on the effective area, A_e, and the operating wavelength. A_e is related to the physical area of the antenna by the aperture efficiency, η. Equation 2.13 denotes this fundamental relationship.

$$G_r = \frac{4\pi Ae}{\lambda^2} \tag{2.13}$$

It is convenient to take the log of both sides of Eq. 2.12 and express everything in decibel units. The term {20 log [4πR/λ]} is used to express the path loss, L_p, in decibels. The path loss accounts for the inverse square decrease of energy as the electromagnetic wave travels to the receiving antenna. The L_p used in Eq. 2.12 is the ideal case and exists when there are no additional losses in the link. R is the distance between the transmitter and the receiver terminals. Equation 2.14 is a more general formula written in logarithmic form that considers all the possible losses associated with the link transmission.

$$P_r = P_t + G_R + G_T - L_P - L_A - L_{EX} \ \ (\mathrm{dBW})(7) \tag{2.14}$$

where,

L_A = losses due to atmospheric absorptions and scattering,

$(L_{EX})_{dB}$ = excess losses due to the system's receiver components and atmospheric losses (ex. rain).

The Friis equation calculates the received signal or carrier power. It is used in the derivation of an essential performance metric known as the Carrier-to-Noise (C/N) ratio.

2.1.4.2 Carrier-to-Noise (C/N)

The link budget is useful in determining our communication system performance (see Figure 2.3). The overall unit of measurement is the C/N factor. We determine an equivalent C/N at the receiver input, which is numerically equal to the true C/N at the detector input. The C/N is typically called the signal-to-noise ratio (SNR) before detection. [15] Equations 2.15 – 2.17 give the C/N formulas for our power budget.

$$P_n = k\, T_s BG \tag{2.15}$$

where,

P_n = noise power at the demodulator input,

k= Boltzmann's constant = 1.38 x 10^{-23} or –228.6 dBW/K/Hz,

T_s = system noise temperature referred to the input of a noiseless receiver,

B_n = narrowest noise bandwidth a head of the demodulator input (usually the IF bandwidth in hertz),

G = overall RF and IF gain of the receiver.

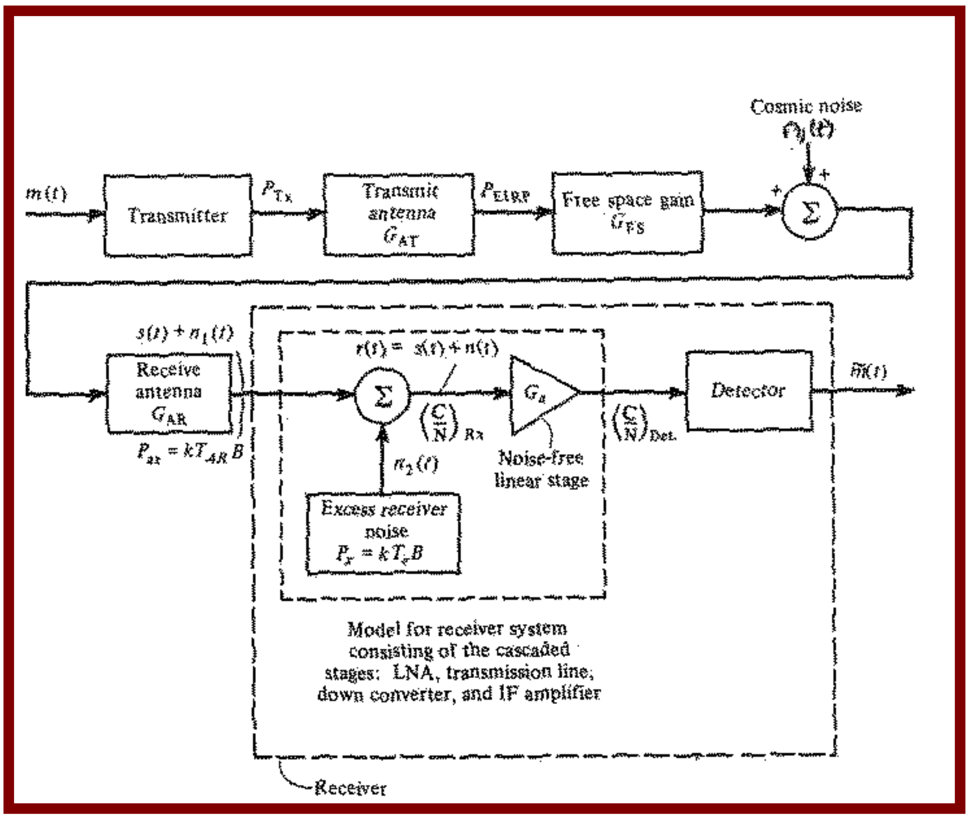

Figure 2.3: Communication System Model for Link Budget Evaluation. [15]

$$\frac{C}{N} = \frac{P_r G}{kT_sBG} = \frac{P_r}{kT_sB} \tag{2.16}$$

Equation 2.17 gives the typically expression for C/N in units of decibels.

$$\left(\frac{C}{N}\right)_{dB} = (P_{EIRP})_{dBw} + (G_{AR})_{dB} - (L_p)_{dB} - (L_{EX})_{dB} - (k)_{dBW/K/Hz} - (T_s)_{dBK} - (B)_{dBHz} \tag{2.17}$$

where,

$(P_{EIRP})_{dBW}$ =10log (P_t)+10log (G_{AT}).

C/N is useful in calculating another important measure of digital communication systems, E_b/N_o. E_b/N_o, which determines the probability of bit error, has a direct relationship with the C/N as explained in the next section.

2.1.4.3 Probability of Bit Error (P_e)

The probability of bit error is used in digital communication systems to describe the quality at the detector output of the recovered data. It is a function of (E_b/N_o). Equations 2.18– 2.19 describe the relationship between E_b/N_o and C/N [15].

$$\frac{C}{N} = \frac{E_b}{N_o}\frac{R_b}{B} \tag{2.18}$$

$$\frac{E_b}{N_o} = \frac{C}{N}\frac{B}{R_b} \tag{2.19}$$

Equation 2.20 describes the equation for the ideal free-space condition of an LMDS communication link operating at 27.5GHz in units of decibels.

$$\left(\frac{E_b}{N_o}\right) = (\mathrm{P_{EIRP}}) + (G_{AR}) - \left(20\log_{10}R\right) + 166.62 - (T_s) - (R_b)\ (\mathrm{dB}) \tag{2.20}$$

where,

R_{dB} =10log(R), R is the data rate (bits/second).

We will use Eq. 2.17 and Eq. 2.20 in evaluating our communication system in Chapter 4.

Having examined those issues relative to the physical layer of the LMDS link, we will now focus our attention towards those issues pertinent to the data link layer.

2.2 Data Link Control (DLC) Layer

This section provides information relative to the data link control layer. The primary function of this layer is to provide reliable data transmission between the LMDS radio ports and our mobile unit's LMDS transceivers. In order to implement this functionality, we will use a fast radio link protocol to insure the continuous flow of data. This layer actually sends blocks of data or frames with the required synchronization, error control, and flow control. We will first discuss error control methods.

2.2.1 Error Correction Methods

Two main techniques exist for reducing errors in digital communication systems, Forward Error Correction (FEC) and Automatic Repeat reQuest (ARQ). The goal of this section is to determine the most suitable methodology for our system architecture.

2.2.1.1 Automatic Repeat reQuests (ARQs)

In the ARQ method, the receiver detects errors in a block of data and then requests retransmission of this data block. Our ideal communication link would allow uncorrupted packet delivery. Realistically, this is not always feasible and the corrupted data must be retransmitted within a certain length of time using the ARQ approach. There are three potential candidates, Stop-And-Wait, Go-Back-N (GBN) or Sliding Window, and Selective Repeat (SR). We will describe the algorithms and list advantages and disadvantages for using each approach.

2.2.1.1.1 Stop-And-Wait

In the Stop-And-Wait routine, the receiver transmits a frame and the sender must wait for an acknowledgment before transmitting the next frame. If the acknowledgement does not arrive within a certain length of time, the transmitter will time out and resend the original frame. When selecting the most appropriate retransmission technique we would like to transmit as much data as possible without waiting for the first acknowledgement. This is not a characteristic of the

Stop-And-Wait approach since the sender must wait until an acknowledgement is actually received before sending more frames. [8] The next approach, Go-Back-N, accomplishes the goal of keeping the pipe full more efficiently by keeping a buffer at the transmitter end of the link.

2.2.1.1.2 Go-Back-N (GBN) or Sliding Window

The concept of Go-Back-N is not complex. Incoming data packets of a transmitting Data Link Control (DLC) module for a link from A to B are numbered sequentially. A sequence number (SN) is sent in the header of the frame at this time containing the packet. The receiving DLC at B accepts the data packets only in the correct order. It then sends request numbers, RN, back to A. [9] One of the inefficiencies of using solely GBN is that when a packet loss occurs, the algorithm is no longer able to keep the pipe full. One solution is to adjust the algorithm to allow early detection of packet losses. This does speed up the process; however, when an error occurs, the protocol must retransmit at least one round-trip delay worth of frames because no buffering was used at the receiver end. A more efficient ARQ that keeps a buffer at both the transmitter and receiver is known as the Selective Repeat.

2.2.1.1.3 Selective Repeat

The Selective Repeat process is identical to the GBN, except when an error occurs, only the block with the error needs to be retransmitted. This functionality occurs because buffers are maintained at both ends of the link to store the uncorrupted frames. The ideal case would be to have infinite buffering. This approach would be advantageous if we have the needed amount of buffering at each end of the communication link.

2.2.1.2 Forward Error Correction (FEC)

Forward Error Correction (FEC) is also known as channel coding because it is used to correct errors caused by channel noise. In FEC systems, the receiver can correct as well as detect errors from the encoded transmitted data. A channel coder encodes a digital message (or source) into a code sequence for channel transmissions. This encoding requires adding redundant bits to the transmitted data stream. These extra bits increase the data rate (bit/s) and the bandwidth of the encoded signal.

There are two classes of FECs: block codes and convolutional codes.

2.2.1.2.1 Block Codes

Block codes are most commonly used for errors that occur in bursts. Block coding involves the mapping of '*k'* input binary symbols into *'n'* output binary symbols. The number of output bits is greater than input bits, which enables the codes to provide redundancy, such as parity bits. These parity bits are added to make "code words" or "code blocks." The block code is known as (n,k) code, and the rate is known as R_c=k/n. This is equal to the information rate divided by the raw channel rate. Parity bits provide error detection and correction at the receiver. The common types of block codes are Hamming codes, Cyclic codes, Reed-Solomon codes, and Golay codes.

2.2.1.2.2 Convolutional Codes

Convolutional codes are different from block codes because information sequences are not grouped into distinct blocks. With convolutional coding, a continuous sequence of data bits is mapped into a continuous sequence of encoder output bits. Passing the information sequence to be transmitted through a linear finite state shift register generates convolutional codes. The shift register typically contains *K* (*k*-bit) stages and *n* function generators. The input data is shifted into the register *k* bits at a time. There are *n* output bits for each *k* bit input data sequence. *K* is the constraint length or the number of input data bits that the current output is dependent upon. The code rate is determined using the same method as block codes. Convolutional coding can be implemented in various ways, such as soft decisions, hard decisions, and Viterbi decoding. [15][4]

2.2.1.3 Fundamental Tradeoffs between ARQ and FEC Methods

Having discussed both methodologies of error correction, we will compare fundamental tradeoffs. ARQ methods are often used in systems that have short transmission delays. They are also used in duplex channels so that the sender can retransmit the required control commands (ex. computer systems). FEC methods are preferred for systems with large transmission delays, such as satellites. They also increase the bandwidth and data rate requirement. The preferred choice is FEC for our system architecture since we are moving at a high-speed and the routing of the ARQ control commands could become complex. An FEC method could detect and correct errors during transmissions. As mentioned earlier, bandwidth is not a major concern at LMDS frequencies and thus will not be a limiting factor when using FEC.

Now that we have discussed methods of insuring reliable transmission, we will consider issues related to the frequency and channel selection.

2.2.2 Duplexing Techniques

In our LMDS network, we desire to have the mobile unit(s) transmitting and receiving simultaneously. This process is called duplexing and can be described as either Frequency Division Duplexing (FDD) or Time Division Duplexing (TDD). In FDD, the duplex channel consists of two simplex channels, one for transmitting and one for receiving. A device called a duplexer will have to be placed in both the mobile unit and the LMDS base station in order to insure simultaneous transmissions and receptions on the duplex channel pair. Regardless of the channel being used in the system, the frequency split between the forward and the reverse channel will be the same throughout a particular base station coverage area.

The advantage to using FDD in our network is that throughput can be increased because both ends of the communication link can use their own separate frequency channel simultaneously. To avoid adjacent channel interference, we must provide enough separation between the frequency allocation. This should not be a major problem with the available bandwidth.

On the other hand, TDD uses time to provide both a forward and reverse link. To create the effect of simultaneous transmission and reception, the time slot between the transmit and receive time must be very small. [4] The disadvantage of this approach is that the high speed of the train limits the bursts duration, thus, preventing simultaneous transmission and reception.

Taking into consideration these fundamental facts, we selected FDD. That is, FDD was selected primarily because of the fast speed of the mobile and the huge amount of information to be sent during each transmission. The following section will examine techniques for multiplexing dissimilar data sources onto an LMDS, FDD channel.

2.2.3 Multiplexing Techniques

Many types of multiplexing technologies exist. Multiplexing is the combination of multiple information channels onto a common high-capacity transmission medium. [11] The primary goal in any multiplexing technique is to use the transmission link (i.e. LMDS) efficiently by multiplexing different types of traffic, such as voice, video, and data, simultaneously. We will discuss four major types of multiplexing techniques: Time Division Multiplexes (TDM), Statistical Packet Multiplexes (SPM), Dynamic TDM, and Asynchronous Transfer Mode (ATM).

TDM involves the assignment of fixed time slots; each is assigned to a traffic channel. Each channel has a fixed bandwidth that is predetermined by the network engineers. The major advantage of using this approach is its ability to guarantee bandwidth for specific traffic types. This concept is quite important for time-sensitive traffic, such as voice and video. The disadvantage of TDM is its tendency to be less useful when the data traffic is "bursty." Once a channel has been assigned, it cannot be used for any other incoming traffic.

SPM is an efficient multiplexing technique that forms the basis of packet, frame, and cell switching. Bandwidth is allocated among multiple stations through the division of time slots. Unlike TDM, the channels are not allocated statically. Bandwidth is allocated only to channels that are passing data at that moment. One of the drawbacks for using SPM is that end-to-end data delivery is unpredictable. This is primarily due to the tagging process, which adds framing overhead, error detection and correction, and retransmission. This time variability makes it unsuitable for voice and video traffic.

Dynamic TDM is a multiplexing technique that combines the best features of both SPM and TDM. The word dynamic originates from the switch between TDM and SPM during the multiplexing process and the method in which channels are made available. For example, three voice calls are initiated on the same 64kbit/s channel. Instead of allocating the full channel, we only use the required bandwidth (i.e. 8kbit/s time slots for each voice call ≈ 24kbit). This would leave a 40kbit/s channel bandwidth to be used by other calls. The 64kbit/s channel will continue TDM multiplexing the voice and data until one of the voice circuits is not required. For instance, if the second call is terminated, then a change of time slots occurs between the third and second slot. The third voice call now moves into the second time slot and we have 48kbit/s available for usage. This dynamic change in multiplexing occurs transparently for both voice and data circuits. [11] ATM is a prime example that leverages from the benefits of Dynamic TDM.

ATM is not just a multiplexing technology, it is also a high-bandwidth, low-delay switching technology. The underlying multiplexing technology is Dynamic TDM. It uses cell relay service (CRS) with a fixed cell size of 53 bytes. It has the basic characteristics of packet switching, but also includes the delay attributes of circuit-switching technology. ATM offers a variety of high-speed digital communication services for transporting voice, video, and data with guaranteed QoS. ATM was chosen as the multiplexing technique for these reasons as well as other important attributes to be discussed later.

This completes the discussion of the data link layer relevant to the proposed system architecture. The following section will discuss issues relative to the network layer.

2.3 Network and Upper Layers Feature

Conceptually, the network layer is the most complex of all the layers discussed thus far since this layer enables all other layers to work together. The network layer is responsible for the routing and flow control functions. In our system architecture, there will be a major need for these functions.

There are two main types of routing techniques: virtual circuits and datagrams. With a virtual circuit, the route for the session is fixed. On the other hand, if datagrams are used, there is not a continuous fixed route for the data packages. In other words, each package from a group leaving the same destination could be routed differently. Also, datagram services is often taken to mean that the network layer can deliver packages out of order (having occasionally dropped packages at the receiving module) and no connection phase is required at the initiation of a session. Again, the virtual connection is the opposite of the datagram services, all packets are delivered once in order, and a connection phase is required at the initiation of the session.

Another major problem at the network layer besides routing is congestion or flow control. Congestion occurs when the users in a network demand more resources than the network has to offer. Both good routing techniques and buffer management at the nodes can help alleviate this problem. The ultimate goal is for the network layer to be able to control the flow of packages into the network. [6] This can be done in our system by having adequate buffers. The buffer sizes could be based on an estimated number of customers and their expected traffic types. The typical file sizes of traffic will be discussed in section 2.4.

The biggest challenge in our network design is how to handle the routing or handover between multiple Infostations or base stations. This section will provide examples of those techniques, such as with the use of ATM or IP. These techniques could possibly play intricate roles in controlling both the flow control and routing in the overall system architecture. In order to determine the transmission protocol, we must understand when to use each of the suggested protocols.

2.3.1 Internet Protocol (IP)

The Internet Protocol layer has actually become known as layer 3.5 or the sublayer that exists between the network and transport layer. Thus, in discussing the role of IP, we must mention TCP, the Transport Control Protocol. IP is an unreliable, connectionless (datagram) delivery service. In other words, it has the traits of the previously described datagrams. IP does make a “special effort” to deliver all packets. The special effort attribute is quite important. It implies that the Internet does not frivolously discard packets. It typically only discards packets when available resources (i.e., bandwidth, buffering) are exhausted or the underlying network fails. Network layer failure could be due to channel conditions at the physical layer and issues at both the data link and network layer.

IP has three main functions. First, it defines the basic unit of data transfer (Internet datagram) used throughout our TCP/IP Internet system. Secondly, it performs the routing function or the selection of the best path to send the data. Thirdly, IP includes a set of rules that characterize how hosts and routers should process packets, how and when error messages should be created, and the conditions on which packets should be discarded. The IP datagram specifies the header format, which includes the source and destination IP addresses, however, it does not specify the format of the data area. Thus, IP can carry various types of data.

The IP datagrams undergo an encapsulation process, which embeds the datagram(s) into a network frame. During the encapsulation process, the physical network treats the entire datagram, including the header, as data. Ideally, we would want to have one IP datagram placed into one physical frame, which would make the transmission across the physical network quite efficient. However, the network hardware sets the upper bound, Maximum Transfer Unit (MTU), on the amount of data that can be transferred in one physical frame. When a datagram has to travel across a network with a smaller MTU than the original MTU, the data packets must be divided into packets known as fragments. Fragments must be reassembled at the ultimate destination. [8]

Thus, the process of encapsulating multiple IP datagram packages is one option for transmitting data in our wireless LMDS network. Another option is to use ATM or a hybrid combination of IP/ATM.

2.3.2 Asynchronous Transfer Mode (ATM)

ATM was discussed in the previous section on multiplexing technologies. ATM is used when there is a need for multiplexing a variety of applications with diverse networking requirements, such as bandwidth and quality of service (QoS) guarantees, over a single physical interface in a high-speed network.

Conceptually, ATM implements the functionality of a network layer protocol; however, it has a protocol reference model similar to the OSI seven-layer model. ATM can be broken into five layers: 1) Physical layer – This layer is responsible for transmission frame generation and recovery, bit transmission capability, line coding, and more. 2) MAC Layer – This layer is optional, but is designed for generic flow control and medium access control functions. 3) ATM Layer –Cell Virtual Path Identifier (VPI)/Virtual Circuit Identifier (VCI) translation occurs at this layer. In addition, the multiplexing and de-multiplexing of cells and header generation and extraction is implemented. 4) AAL Layer – This layer is subdivided into the Convergence and Segmentation and Reassembly (SAR) layers. The main functions are the mapping of higher layer Packet Data Units (PDUs) and the mapping of PDUs to SAR sublayer. 5) Higher Layers – This layer is composed of many other functions in addition to those pertinent to ATM, such as transport protocols (TCP), applications selection, and internetworking protocols (IP). Also, ATM protocols signaling functions are implemented at this layer. [6]

Industry is currently at odds over ATM and IP in the movement toward one common network infrastructure. We will elaborate on this discussion and our choice of protocols in the Chapter 4. Having explored two potential routing protocols, we will next review a discussion on general bandwidth and network requirements for our intended applications.

2.4 Requirements for Real-Time Data Applications

The applications for this system architecture will cover a range of real-time block data applications, such as web browsing, time-sensitive email and file transfer, and voice services. Real-time services require information to be sent immediately. However, non-real time services allow the information to be stored at the receiving party for later consumption, such as non-real time sensitive e-mail and file transfer. Our system architecture focuses on offering real-time application services.

It is far more challenging to provide bandwidth guarantees to real-time block transfer applications due to the frequent change in the size of the Application Data Unit (ADU). In other words, the ADU associated with a particular application can be arbitrary in length and generated at arbitrary instants of time. ADU delay is a challenge that arises from the requirement of meeting the application level Quality of Service (QoS). However, to satisfy the ADU delay requirement, a minimum bandwidth, $\mathbf{B_{min}} = \mathbf{S}$ (size of ADU package in bits) / (the maximum delay) $\mathbf{D_b}$, can be determined with the assumption that the application can be continuously sent at the rate B_{min} in order to deliver the ADU at a constant bit rate. The ultimate goal is to insure that the last packet of the ADU arrives before D_b. We can determine the traffic requirements for this class of service by the ADU size and the ADU delay requirements. Table 2.6 gives a list of common real-time block transfer applications with the typical values for the aforementioned parameters. [6]

Table 2.6: Examples of Bandwidth Requirements of Real-Time Block Transfer Application

Applications	ADU Delay (D_b)	ADU	ADU size (S)	B_{min}
Web browsing	100ms	Typical Web object	3kbytes	240kbps
	100ms	Large Web object	20kbytes	1.6Mbps
File Transfer	1min	Large software application	10Mbytes	1.3Mbps
Chat	1sec	Words	100bytes	.8kbps

Although voice requirements are not shown in Table 2.6, the general total round-trip delay time should not exceed 500ms in order to achieve acceptable quality for a telephone conversation. (i.e. average 250ms one-way transmission). [5] Meeting delay constraints is highly important in our depicted environment in order to provide a continuous seamless data flow. These parameters and others are essential components in designing efficient handoff, routing, and flow control mechanisms.

2.5 Summary

Chapter 2 has provided key suggestions for modulation selection, error correction coding, multiplexing techniques and duplexing techniques. Networking issues were also discussed. Parameter selections were made which catered to the requirements of offering continuous packet services via a robust high-speed LMDS environment. These parameter selections will be used in all of the proposed system architectures. The next chapter provides a set of system specifications, which uses these parameters in developing a baseline for system throughput requirements.

Chapter 3

System Specifications

This chapter provides the required specifications for our mobile LMDS high-speed system architectures in terms of throughput data rate requirement. Our design goal is to serve many customers with varied real-time applications.

3.1 Raw Payload Requirements

Our focus customers are high-speed train commuters. The most recently built high-speed train, which travels at 150mph, is the *Amtrak Acela Express*. The Acela Express capacity consists of Bombardier-built train sets with a maximum capacity of 304 passengers. [18] We will initially design our system to serve approximately one-third (i.e. 100) of this maximum capacity simultaneously. Table 3.1 shows the distribution of services and the raw payload requirement. Table 2.6 provided the benchmark for the information in Table 3.1.

Table 3.1: Characteristics of Offered Services/User

Applications	**ADU**	**ADU size(S)**
Web browsing 3 objects/page; 1 page/user	Typical Web object (3kbytes)	9kbytes
File Transfer	File Average Request	.15 Mbytes
Chat/Email	Words (short email session)	1kbytes
Voice/POTS	Voice Channel/user	6.32Mbps (DS-2) supports 96 voice channels or 8kbyte/user
Total Payload/User		.168Mbyte or1.344Mbps
Total Required System Payload for 100 users		**134.4Mbps**

The voice packets data rate in Table 3.1 was chosen based on the standard T1 rates. If each user has a single dedicated channel, then the total data rate needed to support 100 users is 64kbps * 100 (i.e. 6.4Mbps). This amount is relatively close to the common carrier T2 rate that supports 96 channels at 6.312Mbps. Thus, we used the common carrier rate of 6.312Mbps.

The proposed system will need to deliver a raw payload of 134.4Mbps having a maximum BER of $10e^{-6}$. On the train, we will demultiplex this to support 100 users, each getting 1.344Mbps. However, additional overhead will be added due to channel coding and clock synchronization as discussed in the following section.

3.2 Overhead Requirements

We will discuss the system requirements for both downstream and upstream traffic. Table 3.2 gives a description of the downstream parameters, such as overhead and required transmission rate to support our train commuters. An explanation for the italicized parameters will be given as well.

Table 3.2: Downstream Payload System Requirements

Description	**Downstream (To Users)**
Modulation	N-FSK
Level	4
BER before un-coded	$10e^{-6}$
E_b/N_o Required	10.76 (un-coded), see Table 2.5
Convolutional Coding	R_c= 2/3, coding rate
Bit Interleaving	M = 2
Overhead for Routing	.597ms
Overhead for clock synchronization (includes frame and bit timing)	Negligible Contribution
Required Transmission Rate or Channel Bit Rate with FEC and overhead	201.51Mbps

Below is a brief description of the parameters chosen in Table 3.2:

- **Convolutional Coding**. We selected convolutional coding because it provides a continuous process of encoding and decoding for data streams. It is better suited to a moving train and a system designed to offer continuous communications than block coding, which is primarily used in bursty environments. In choosing the best code rate, we must consider the amount of payload processing time and the available bandwidth. With these parameters, we selected a code rate of 2/3 for both the downlink and uplink channel. This implies that the required data rate from FEC overhead will be (R_b*n/k) or 133.4Mbps*1.5 ≈ 200Mbps for the raw payload.

- **Bit Interleaving**. We selected bit interleaving to reduce errors that occur in bursts. Our burst errors are more likely to occur during periods of rain fades or due to a sudden obstruction in our LOS path. In addition to increasing the power level, we could also perform a bit interleaving process to guard against these random error bursts. An interleaving of degree m has m rows (code words), which creates an interleaved code word of (m*n, m*k). [15] We will set our value of m to be small since this process impacts processing time.

- **Dating Routing Overhead.** The data routing calculations refer to the total time required to route the data from the source network to the LMDS base stations. We think of five major delay components in an end-to-end network for packet data units (PDUs):

  1) *Packetization delay* is not an issue in our design. It is typically only incurred when delivering live encoded source for real-time applications.

  2) *Transmission delay* (per PDU) refers to the time it takes to transmit one PDU from the source to destination. It is dependent on the PDU size and transmission rate of the local link. We can determine the transmission delay by dividing the PDU by our assumed transmission rate, 155Mbps. We will assume a PDU of 500 bytes. It would take one PDU a total of ((500*8)/155Mbits) = 26us.

  3) *Propagation delay* (per PDU) refers to the amount of time it takes a PDU to propagate through the medium over a certain distance. Reference 6 provides a unit metric that indicates that for every 200m, the propagation delay is 1us. We will assume that the

maximum propagation distance is 5km, which implies maximum propagation delay of (5km/200) * 1us = 25us/PDU.

4) *Queuing delay* (per packet) is a major issue in a packet switched network because it deals with the total buffering delay incurred at every packet switch along the routing path of the packet PDU. Typically, voice packets travel along circuit switched networks. In addition, queuing delay depends upon the arrival rate of the packets to the queue and the current capacity load of the queue. We will make many assumptions for simplicity. Assume that each packet travels along one OC-3 link (in reality, this could vary between 28.8kbps to OC-3 or higher) and three router hops. Each router will have an average of 10 packets queue length, using an average packet length of 200 bytes (typical for Internet). The average queuing delay on this path would be ((10*200*8)/155Mbps)) * 3 routers =.516ms. This calculation is an example of a worst case scenario.

5) *Processing delay* is the total processing time needed at each switch and end system. This is typically negligible compared to the queuing delay. The typical process time for an ATM switch is 10µs. The switching speed varies depending on the current network condition. We will assume the packet travels through three ATM switches, thus the total processing delay is 30us.

We can now approximate the aggregate routing delay as (30us + 25us + 26us + .516ms) = .597ms.

- **Overhead for clock synchronization (includes frame and bit timing).** We have discussed both bit and frame (word) synchronization previously. The actual bit synchronization needed depends on the line coding. For simplicity, we will assume a unipolar RZ line coding with a sufficient number of alternating 1s and 0s in the data. With this assumption, our bit synchronization time is inversely proportional to the inverse of our bit rate, 1/(133.4Mbps) = 7.496e-9s/bit. We will assume that each frame consists of 188 bytes (see Reference 7). Thus, the total frames in our payload equals (133.4Mbits/(188*8)) = 88896. This implies that the total frame synchronization for the raw payload is (88896

*7.496e-9) = .664ms. This assumes that framing synchronization occurs during one bit period.

In computing the total data rate for transmission, including the raw payload and overhead, we first recall the 200Mbps required for error correction coding. Bit synchronization is (1/(200Mbps) = 5e-09 s/bit. Recall, the total overhead for frame synchronization is .664ms. Thus, the total channel bit rate, including frame synchronization, data routing, and error correction coding is 201.51 Mbps. We will now analyze the bandwidth allocation for both upstream and downstream based on this required data rate specification.

3.3 Upstream and Downstream Bandwidth Allocation

LMDS has approximately 1.3GHz bandwidth available for both downstream and upstream transmissions. The downstream (from LMDS base station to user) will be greater than upstream due to the pre-determined payload of each user. Using Carson's rule (see Eqs. 3.1 – 3.2), ΔF must be ≤ $\frac{B_T}{2}$. Thus, 2ΔF must be large enough so that the spectra of the four signals have negligible overlap. We will arbitrarily assign ΔF to be 500kHz. Equation 3.2 gives the total needed transmitted bandwidth (B_T). According to Table 3.2, our modulation level (M) is 4. Modulation level M=4 has a 1-to-1 efficiency ratio. As previously discussed, the higher the modulation level, the greater bandwidth required for FSK (see Table 2.5).

$$B_T = 2\Delta F + (1+r)\frac{MRb}{2\log 2M} + 2fd\ (fd\ @120mph) \tag{3.1}$$

$$B_T = 2(500kHz) + (1+.3)\frac{4*201.51e6}{2\log_2 4} + 2(4878Hz) = 263MHz \tag{3.2}$$

The results from Eq. 3.2 reveal that we need to allocate approximately 300MHz for the downstream channel. This implies that we will have an excess bandwidth of (850MHz – 300MHz) = 550MHz. Hence, we can afford to allocate the same amount of bandwidth in both directions. However, we will only use one-half of the transmitted bandwidth (150MHz). It is assumed that the traffic upstream will be much less because the users will only send requests as opposed to the downstream when the actual payload is transmitted to the users.

Figure 3.1 shows the bandwidth allocation for both upstream and downstream transmissions. It also shows the placement of the pilot tones. Pilot tones have various functions, such as gain regulating, monitoring, frequency comparison, and measuring stability levels. A diversity combiner for sensing pre-detection continuity of signals could be used to insert the pilot tone. That is, the purpose of the pilot tone is to tell the combiner that the transmission path is a valid one. [19] It is inserted prior to modulation to monitor the reception of the receiving mobile unit. We desire to place the pilot tone in a location where it will not interfere with the signal spectrum for either downstream or upstream. Because of this, we will place it at 27.65GHz between the upstream and downstream channels. The marginal separation between the upstream and downstream channel will be 50MHz.

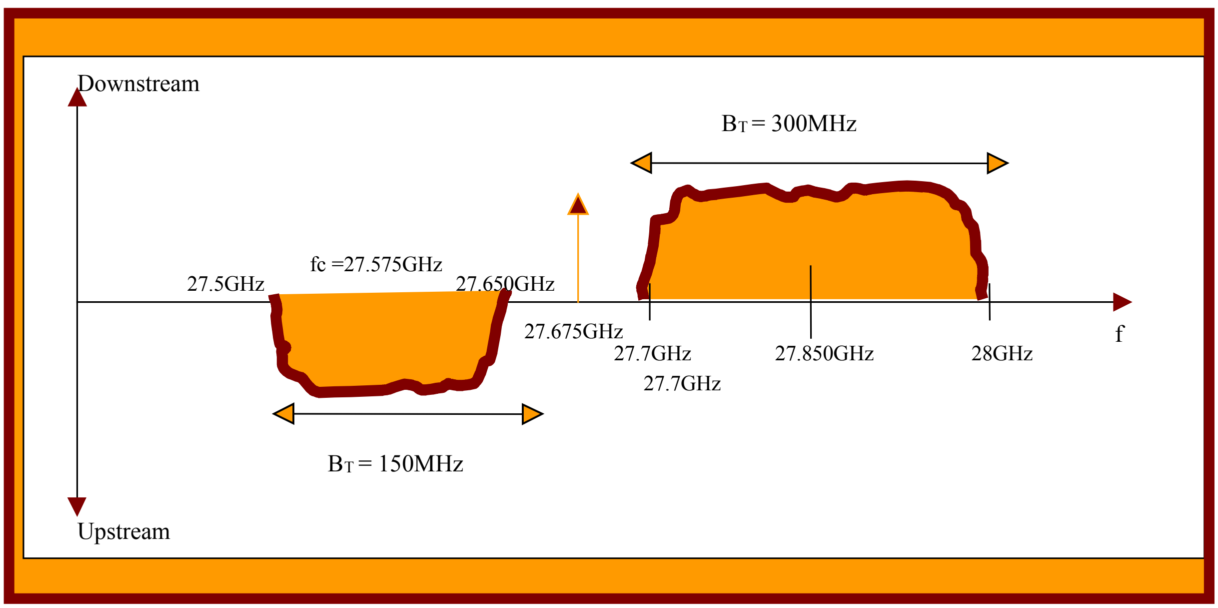

Figure 3.1: Spectrum Allocation for Upstream and DownStream Channel

With the large available bandwidth remaining, the following is a list of other alternatives we could enforce to utilize the excess bandwidth: 1) offer more services, 2) increase the amount of services currently being provided, 3) increase the number of customers, 4) increase the channel coding rate, 5) increase the frequency separation among symbols, and 6) increase the modulation level. However, with the required bit rate, there is a cap on the permissible modulation level.

We next compute the implications of using a larger modulation level (M=64). From Eq. 3.1, we see that the bandwidth requirement is beyond the capacity of the total allocated bandwidth (1.3GHz). Using M=64 is feasible if we decrease the number of customers, which in turn decreases the required throughput. The minimum throughput that could be supported, including all overhead requirements, would be approximately 144Mbps. This assumes only 1GHz of available bandwidth for downstream and the other specifications remain constant.

$$B_T = 2(500KHz) + (1+.3)\frac{64 * 201.51e6}{2\log_2 64} + 2(4878Hz) \approx 1.4GHz$$

3.4 Summary

This chapter has provided an overview of system throughput requirements. We use bandwidth requirement for our broadband services and associated overhead for routing and convolution coding to determine this required throughput. We also discussed the use of pilot tones in our upstream and downstream bandwidth allocation.

The following chapter will use this calculated throughput as a part of the criteria for selecting the best methodology for our high-data rate LMDS mobile network.

Chapter 4

LMDS Mobile System Architectures

Our ultimate goal to provide continuous real-time services is the focus of the system architecture. This chapter provides a thorough analysis of three proposed designs. The chapter begins with an introductory discussion of the LMDS Infostations approach. We will next explore link budget scenarios and the impact of these parameters on our architecture. The physical layout is also discussed relative to the railroad track architecture. The succeeding sections provide similar information analyzing the other approaches, Tower Sites and Infostations/Tower Sites. Networking issues will be briefly discussed. Finally, a summary of the best approach will be provided.

4.1 Proposed Approaches

4.1.1 LMDS Infostations

Infostations are wireless cells (i.e. LMDS base stations and their coverage areas in our system architecture) designed to offer individual pockets of high bandwidth connectivity for broadband data and network services. Their distributed, isolated nature and internal networking structure enables them to operate as stand alone wireless data sources in remote locations and to deliver huge amounts of information during one transmission burst.

The concept of Infostations originated at the Wireless Information Network LABoratory (WINLAB) as wireless systems designed to provide "anytime anywhere" terrestrial communication services. Infostations are particularly suited to high bandwidth mobile services.

[2] The "anytime anywhere" cliché refers to the ability to access applications in a "drive through," "walk-through," or "sit-through" environment. The base stations are called LMDS Infostation Radio Ports (LIRPs) and the LMDS Mobile Units (LMUs) are the customer premise equipment. Our network design will best suit the "drive through" scenario because the LMDS Mobile Units (LMUs) will be near a particular LIRP only for a short amount of time. The next section will offer a visual perspective of the overall system architecture.

4.1.1.1 Architecture

The LMDS Infostations Approach is illustrated in Figure 4.1. Multiple narrow beam antennas are simulcasting broadband data to a single fan beam antenna on the train. The transmission periods are dependent on the coverage area of the transmitting antennas. We observe base station (I) in a dormant mode because it has just completed data transmission with the train. Base station (P) is sending pilot tones in attempt to establish communications. The separation between theses base stations and the choice of antennas will be discussed in more detail in the following sections.

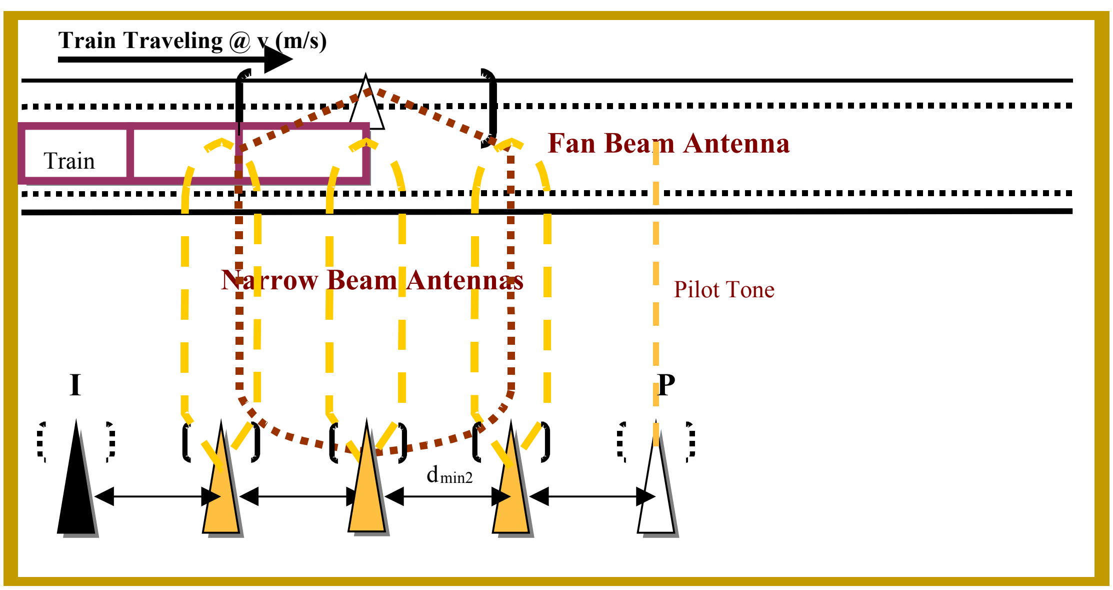

Figure 4.1 LMDS Infostations Architecture

4.1.1.2 Operations

We will briefly summarize operational recommendations as provided in Chapter 2 relative to the LMDS Infostations (LI) approach. Communications between the LMDS Infostations (LIs) and LMDS Mobile Units (LMUs) is initiated with pilot tones. Upon connection, a large amount of information is transmitted within a short burst. FDD is used to transmit data on two separate frequency channels. The selected modulation is non-coherent M-ary FSK since its lacks phase sensitivity and it is robust in our working environment. The error correction will be implemented using an FEC method. At the end of each transmission burst, we will have a dormant period, which limits the amount of transmitted information. This process is repeated for the duration of the mobile connection to our LMDS Infostation Network. With any wireless system architecture, it is important to analyze the required link budget parameters as discussed in the following section.

4.1.1.3 Link Budget Analysis

An important step in the Infostations demodulation process is the acquisition of our original transmitted bit streams. One of the keys to extracting the transmitted information is obtaining the required E_b/N_o and C/N at the receiver input. An extensive study was implemented on the link budget parameters required in obtaining these performance metrics. We will use the link budget equations as discussed in Chapter 2.

The high-speed mobile environment and intended application services were contributing factors to the parameters for our LMDS Infostations link budgets. An explanation for each parameter is given immediately following the link budget table. Tables 4.1 and 4.2 provide tradeoff considerations for our link budget. All link calculations were implemented with a design goal of P_b equal $10e^{-6}$.

Table 4.1: Link Budget for LMDS Infostations Network

Parameters	**Intermediate Parameters**	**Values**
P_t	50mW	17 dBm = -13d BW
$\mathbf{G_{AT}}$ (Pyramidal Horn)	H-θ_{3dB} = 20° E-θ_{3dB} = 18°	18.4 dB
$\mathbf{G_{AR}}$ (Fan Beam)	H-θ_{3dB} = 80° E-θ_{3dB} = 40°	10 dBi
$\mathbf{L_p}$	R = 5.7m	77 dB
$\mathbf{L_{EX}}$		5 dB
$\mathbf{T_s}$	600K	27.8 dB
k (constant**)**	1.38×10^{-23} J/K	-228.6 dB
$\mathbf{R_b}$	155Mbps (OC-3)	82 dB
$\mathbf{B_n}$	80MHz	79 dB
$\mathbf{I_m}$		2 dB
C/N calculated		**50.2 dB**
$\mathbf{E_b/N_o}$ calculated		**53.2 dB**
$\mathbf{E_s/N_o}$ required	4 FSK	**13.8 dB**
$\mathbf{E_b/N_o}$ required	4 FSK	**10.8 dB**
$\mathbf{E_b/N_o}$ margin		**53.2 dB – 10.8 dB = 42.4 dB**

An explanation for each parameter is as follows:

- $\mathbf{P_t}$, transmitted power, was chosen to be a low value in order to limit the amount of power used during favorable weather conditions. This value would increase during high rain periods or other propagation link impairments. However, the exceptional large link margin gives provision for link impairments in the system design. Another reason for using a low value was to decrease the design cost of the LMDS transmitter and/or transceivers. In addition, one of the essential characteristics of using the Infostations approach is its ability to operate using low power components.

- $\mathbf{G_t}$**,** transmitter antenna gain, was derived using formulas of a pyramidal horn (see Equations 4.1– 4.3). [13] We selected a horn antenna for the LMDS base stations because it is known to provide high gain for relativity wide bandwidths at microwave regions above 1GHz. In addition, the theoretical gain of a horn antenna is practically attainable. The pyramidal horn has dimensions in both E and H-planes. This design leads to a narrow beamwidth in both principal plans, instead of individual E or H plans as provided with sectoral horn antennas. [14] This antenna design was chosen because of the directivity gained and needed during short bursty transmissions.

$$\theta_{3dB} = \frac{70\lambda}{d_H}(\text{H plane - horizontal}) \tag{4.1}$$

$$\theta_{3dB} = \frac{56\lambda}{d_E}(\text{E plane - vertical}) \tag{4.2}$$

$$Gt = \eta\frac{4\pi d_E d_H}{\lambda^2}, \eta = .51\ (\text{optimum}) \tag{4.3}$$

The 3-dB beamwidth (θ_{3dB}) or half-power beamwidth is the typical value used to measure the power level for antennas. It is the angular width of the main beam of the antenna radiation pattern located 3-dB below the main-beam peak. The main beam is the lobe containing the direction of maximum radiation. At the extreme opposite end of the main lobe, we have the beamwidth between first nulls (BWFN). BWFN is approximately 2.5 times the 3-dB beamwidth. Figure 4.2 shows the typical power radiation polar plot of this relationship.

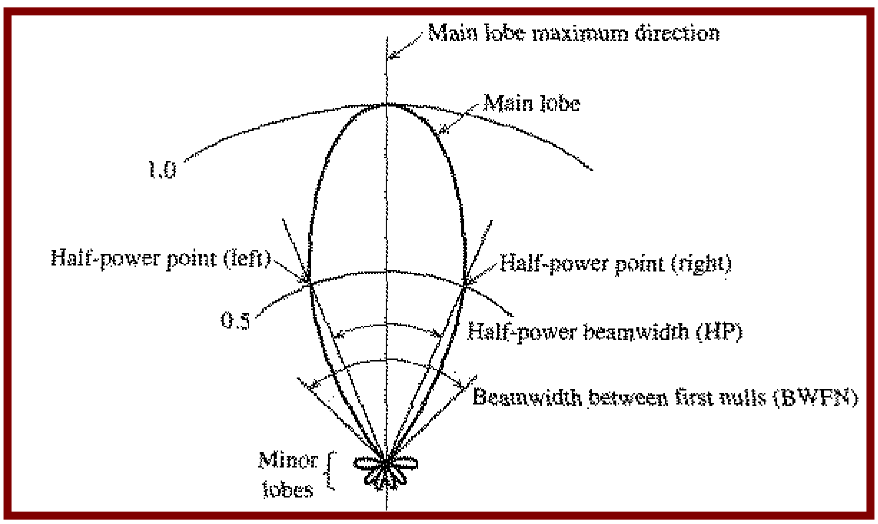

Figure 4.2: A Typical Power Pattern Polar Plot [14]

- **G_{AR},** receiving antenna gain, of the broadband fan antenna was derived using an approximate directivity equation used to give an estimated gain (see Eq. 4.4). [20] Fan-beam antennas have narrow beamwidths in one plane and a much broader beamwidth in the orthogonal plane. Figure 4.2 is a type of fan-beam antenna. A fan-beam antenna was chosen in order to increase the beam coverage area, which in turn decreases the number of antennas needed on the train. In addition, this antenna selection will allow multiple transmitting antennas to communicate to a mobile unit(s) with a pre-determined distance. That is, mobile units will be able to transmit to one receiving antenna within a given range. This range and number of transmitting antennas will be evaluated in later sections.

$$D \approx \frac{32{,}400}{\theta_1 \theta_2} \tag{4.4}$$

- **L_p**, path loss, was determined by considering the average maximum right-of-way distance, R, of typical high-speed trains. This approximated value is eight feet (2.5m). If we decide to place the LMDS Mobile Units on the opposite side of the train, then we must also consider the width of a train car. The average value is 3197-mm (3.2m). Thus, R is equal to the right-of-way distance plus the train width or 5.7m. The path loss formula is given in section 2.1.3.1. Another consideration is multiple tracks. If we have two trains on separate tracks, then R will increase by the distance between the two tracks and the width of the second track.

- **L_{EX}** refers to excess loss other than free space. This value was determined by assuming losses, such as attenuation in atmosphere and those associated with the transmitting and receiving antennas.

- **T_s**, system noise temperature, is a useful value in communications systems. This value depends on the noise temperature of the antenna and the receiver components before the detector input. It can be approximated using Eq. 4.5.

$$T_{system} \approx T_{antenna} + T_{LNA} \tag{4.5}$$

This approximated equation can be used if the low noise amplifier (LNA) gain is high enough to reduce the effect of later stages in the receiver to a negligible level. However, in certain systems we consider the temperature generated from all receiver components before the detector. The typical temperature value for terrestrial antennas is 63°F or 290K. Thus, the receiver components contribute to a large variation in T_s.

- **R_b** is the data rate of the system. We chose this value because we would like to provide our users with a large amount of data given our small transmission time. The amount of data transmitted at a particular bit rate is highly determined by the speed of the train and the characteristics of the antennas.

- **B_n** is the noise bandwidth as discussed in Chapter 2. According to [7], when we do not know the equivalent noise bandwidth, we use the 3-dB bandwidth of our receiving system.

- **I_m** is the implementation margin. It is commonly 2dB for data rates greater than 100Mbps. Typically, systems do not operate under ideal conditions using perfect receivers. Thus, implementation margins are required to compensate for the invariability of filters, clock jitter, and fluctuations in digital receivers. This adjustment is reflected by a reduction in the E_b/N_o and C/N calculated values.

- **C/N calculated** was determined from Eq. 2.17.

- **E_s/N_o and E_b/N_o required** were chosen from Table 2.5 using 4-ary FSK. Equation 2.8 gives the relationship between E_s/N_o and E_b/N_o. We must note that the values in Tables 2-4 - 2.5 assume we are using ideal Nyquist pulse shaping in an additive white Guassian noise (AWGN) channel.

- **E_b/N_o calculated** was determined from Eq. 2.20. Recall, Equations 2.18 - 2.20 give the relationship between E_b/N_o and C/N.

- **E_b/N_o margin** was determined by subtracting the theoretical E_b/N_o of 4-FSK from the calculated E_b/N_o. Recall, the calculated E_b/N_o includes the implementation margin.

Table 4.2 provides maximum selection values resulting from a worst case implementation. It also attempts to decrease the large margin in E_b/N_o. Link parameters that were changed from Table 4.1 are italicized.

Table 4.2: Link Budget for LMDS Infostations Network, Scenario 2

Parameters	**Intermediate Parameters**	**Values**
P_t	1mW	0 dBm = -30d BW
G_{AT} (Pyramidal Horn)	H-θ_{3dB} = 20° E -θ_{3dB} = 18°	20 dB
G_{AR} (Fan Beam)	H -θ_{3dB} = 80° E -θ_{3dB} = 30°	13.5 dBi
L_p	R = 5.7m	77 dB
L_{EX}		7 dB
T_s	900K	30 dB
k (constant)	$1.38x10^{-23}$ J/K	-228.6 dB
R_b	850Mbps	89 dB
B_n	450MHz	86.5 dB
I_m		2 dB
C/N calculated		**27 dB**
E_b/N_o calculated		**29.6 dB**
E_s/N_o required	64 FSK	**14.8 dB**
E_b/N_o required	64 FSK	**7 dB**
E_b/N_o margin		**29.6 - 7 dB = 22.6 dB**

Table 4.2 results for C/N and E_b/N_o decreased drastically as desired. We noticed that even when using 64-FSK, the required E_b/N_o is still attainable with a sufficient link margin. We also noticed that our goal could be attained even with a massive decrease in power level or significantly more attenuation. The following section uses the pre-selected link budget parameters to compute the system's throughput.

4.1.1.4 Performance Evaluation for LMDS Infostations

This section provides a discussion on determining the overall throughput of the system using the proposed link budget parameters. The maximum amount of data distributed during a communication burst is determined by the beamwidth of the transmitting antenna, the distance R, and the speed of the vehicle. Equations (4.6 – 4.9) provide this relationship.

$$C_d = (\theta_{3dB} * R) \tag{4.6}$$

where,

R = average maximum right-of-way distance of high-speed trains,

C_d = coverage arc length (m).

The coverage area assumes a track geometry as shown in Figure 4.3.

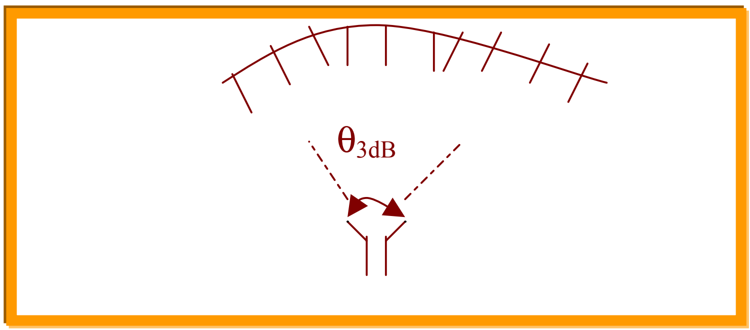

Figure 4.3: Track Geometry using the Pyramidal Horn Antenna

$$V = \frac{C_d}{T} \tag{4.7}$$

where,

T = duration the train travels in seconds,

V = velocity of the train (m/s).

$$R_b = \frac{D_b}{T} \tag{4.8}$$

where,

D_b = amount of data transmitted in bits during T.

$$D_b = \left(\frac{(\theta_{3dB} * R) R_b}{V} \right) \tag{4.9}$$

Figure 4.4 shows the performance curves of throughput–vs-speed. We used a constant value of θ_{3dB} from Table 4.1 of 20°. R_b was compared with the required bit rate, 202Mbps, and the common OC-3 link capacity. Given the beamwidth constraints, the results reveal that the bit rate must be at least 200Mbps in order to offer continuous service with the given speeds.

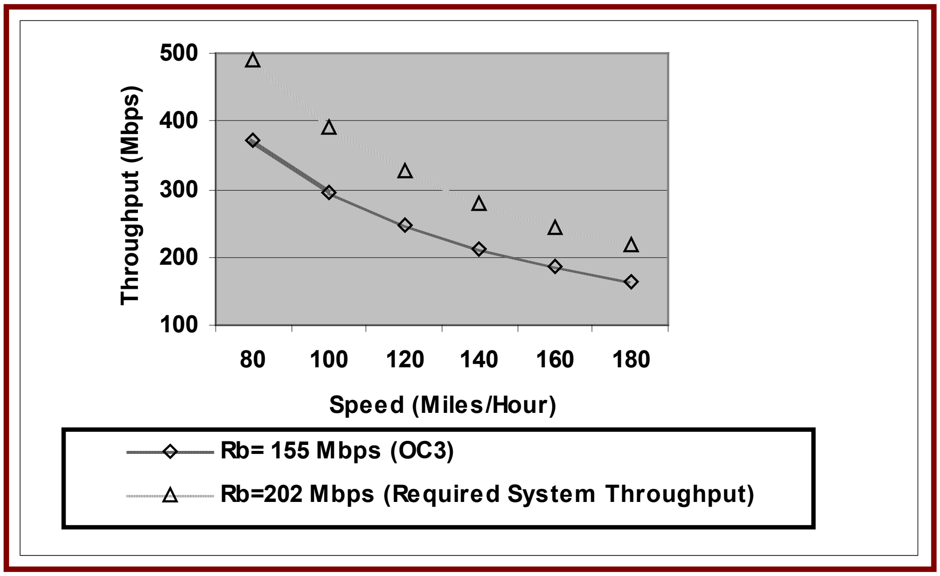

Figure 4.4: Throughput as a Function of Bit Rate (R_b)

Figure 4.5 shows a change in throughput as a function of the transmitting antenna beamwidth. The bit rate (202Mbps) and distance R (5.7m) remain constant. As the beamwidth increases, we observe an increase in throughput as expected. We only have to use a 15° beamwidth for the transmitting antenna in order to guarantee throughput of the required bit rate (202Mbps)

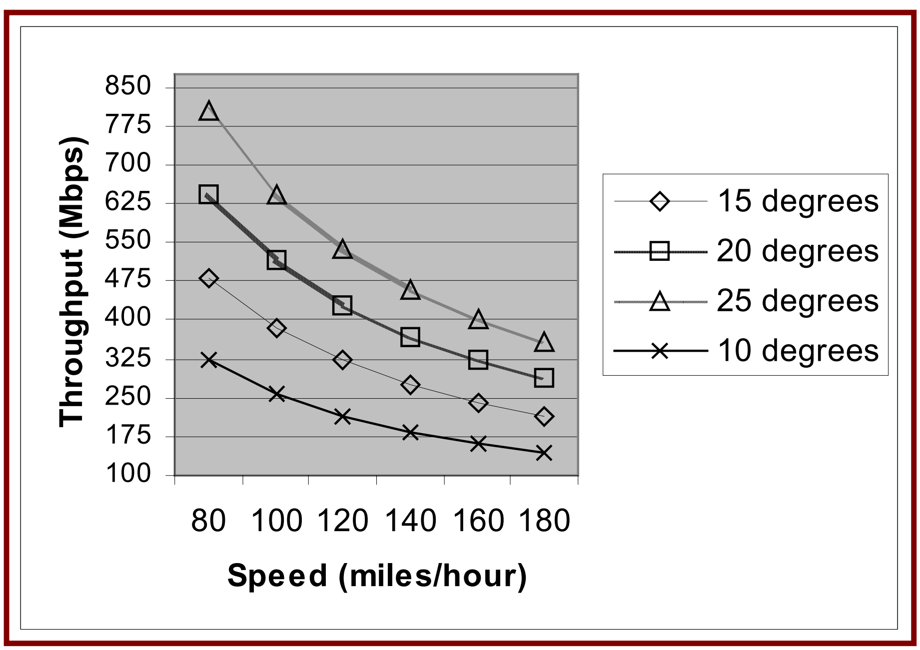

Figure 4.5: Throughput as a Function of Transmitting Antenna (θ_{3dB})

The above throughput results will not be feasible without the proper layout of the LMDS Infostations and mobile units. The next section discusses a deployable layout using both our link budget parameters and a typical track layout design.

4.1.1.4.1 Minimum Distance Separation

Our next design goal is to determine the minimum distance between two adjacent Infostations needed to provide continuous services. There are two scenarios to consider. First, the minimum distance separation could be computed by the width of the H-plane beamwidth on the transmitting antenna (e.g. 20 degrees). Equation 4.10, similar to Equation 4.6, provides the equation for the minimum distance, d in meters. This equation assumes that the time needed to route the data packets to the appropriate user is negligible.

$$d_{\min} = \theta_{3dB.TX} * R \qquad (4.10)$$

Another equation considers the minimum acceptable delay between voice packets. This delay was considered because voice applications have critical requirements. As alluded to in Chapter 2, the maximum one-way voice delay is 250ms. However, other studies have given the value of quality voice calls delay to be considerably less than 150ms. [6] In order to meet this delay, we will consider a value of 125ms. This value assumes that the voice packets have a higher priority than data packets. The new distance, d_{min2}, between base stations can now be determined by using Eq. 4.11.

$$d_{\min 2} = d_{\min} + 125ms * V \qquad (4.11)$$

Equation 4.11 reveals the dependence on the speed of the train on the separation between adjacent base stations. We will use these equations to calculate the total number of LMDS Infostations. This computation is the subject of the following section.

4.1.1.4.2 Track Layout

We will consider two layout designs for the LMDS Infostation approach. One case examines the train track with the turning radius of curves. Ideally, the best case would have a straight track with radius of curvature equal zero. Figure 4.1 provides a visual perception of the LMDS

Infostation methodology. The figure is not drawn to scale. However, the fact that only one broadband-receiving antenna is required to serve multiple base stations is an important observation. This is only true when providing continuous service based on the limitation of d_{min2}.

We will first derive the number of needed terminals using Eq. 4.12.

$$n = \frac{D_t}{d_{\min}} \tag{4.12}$$

where,

D_t = total length of the train in meters.

If we assume a straight route of 230 miles and d_{min} equals (20°*5.7m), then n = 3247 required stations. Figure 4.6 takes advantage of the delay value discussed in Eq. 4.11 and plots the speed versus the number of required stations. Note that even if the train is traveling at 80mph, the number of required stations is less than the first approach. However, with either approach, the number of needed mobile units (broadband antennas) is one. If the coverage area of the receiving beam is 80*5.7 (456m) and the typical length of the train is 160 meters, then only one fan beam antenna is required to provide continuous service. This antenna could service $(456/(d_{min}))$ = four LMDS base stations.

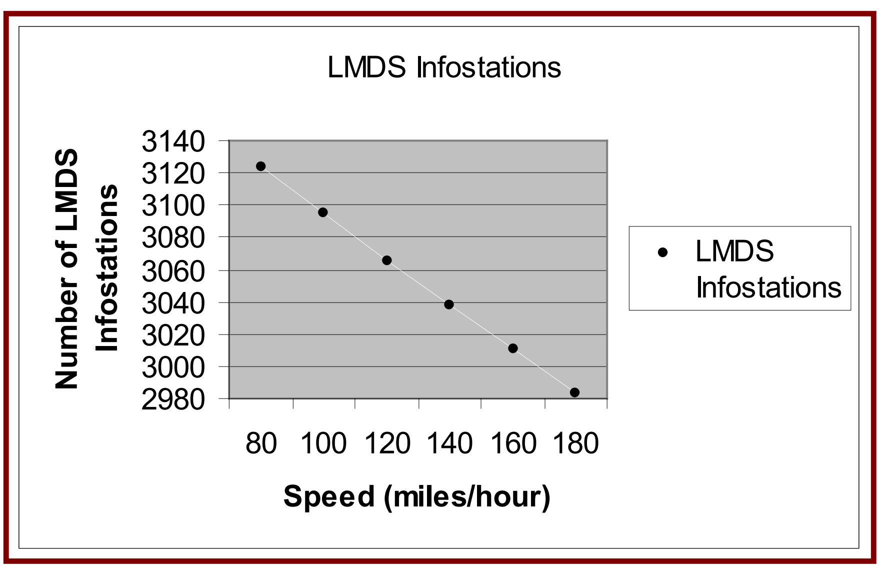

Figure 4.6: Speed of Mobile Units -vs- Number of LMDS Infostations

We will next consider a more realistic scenario with curves of varying radii along a railroad track. Table 4.3 gives typical radius values for high-speed trains across various countries. This table also gives the track length and design speed of the train. Length of curve and depth of radius are factors in the actual deployment of the base stations. To offer full continuos service we wish to design the layout so that the base stations will be placed at the end of curves or at d_{min} away from the end of a curve. These approaches would allow us to supply full, continuous coverage. Depending on the number of curves and their location on the track, we might have to provide multiple LIs within the same curvature area.

Table 4.3: Typical Radius of Curvature Statistics for High-Speed Trains [22]

Country	**France**	**Germany**	**Italy**	**France**	**Spain**
Line	Paris-Lyons	Hannover-Wurzburg	Rome-Florence	Paris-Bordeaux	Madrid-Barcelona
Track Length	427km/ 265 miles	327km/ 203 miles	260km/ 161 miles	260km/ 16 1 miles	522km/ 324 miles
Design Speed, V_{max} (km/h)	300 or 186m/h	250 or 155m/h	250 or 155m/h	300 or 186m/h	300 or 186m/h
Radius of curvature, R_{min}(m)	4000	7000	3000	4000	4000

In evaluating the layout of our system architecture, we must also consider the height of the LMDS Infostations and the LMDS Mobile Unit on the train. These dimensions are determined by the height of the train tunnels, which are designed based on the train and track layout.

A typical total height for high-speed locomotives is 3m (L_H). The railroad track height must be considered and will be denoted as R_H (ex. 200mm). T_H will be used for the tunnel height. Measurements of typical tunnel dimensions were taken at the Montgomery West End Tunnel in Blacksburg, VA. T_H was measured to be approximately 15-ft (4.57m). The actual tunnel length varies depending on the geographic location. With these factors, we can develop a formula for estimating the height of our units. Equation 4.13 gives an approximate maximum height of the mobile units on-board the train. This equation results in the maximum clearance of the tunnel.

$$LM_H = T_H - (R_H + L_H) \tag{4.13}$$

Using our known approximate values, we get

$$LM_H = 4.572 - (.2m + 3m) = 1.372m$$

Equation 4.14 gives the approximation for the height of the LMDS Infostations (LI).

$$LI_H = LM_H + (R_H + L_H) \tag{4.14}$$

Substituting equation 4.13, we see that the maximum height of the LIs is equal to the height of the tunnel as expected.

Another important factor to consider is the width of the tunnels. The width must be large enough to provide adequate space for the deployment of our LIs. The total width, T_W, was measured to be 14′5″ (4.4m). The width between the train tracks and the side of the tunnel was 5′5″ (1.65m). Considering these parameters and the typical width of high-speed trains, 3.2m, we can assume that adequate space is available for placement of base stations.

However, if technical or economical reasons prevent the placement of base stations inside the tunnels, a message could be broadcast to the current mobile users or along the information panels in the train to indicate a temporary outage. If this occurs, we would design our system to track the location of a particular user data session. This would allow a continuation of the service whether than restarting the entire session.

We have provided a detailed discussion on the physical system architecture for a mobile LMDS system using the Infostation approach. A summary is provided in the next section with an introduction to a new methodology.

4.1.1.5 Summary

We can conclude that the large number of LMDS Infostations required to support continuous services would not be economically efficient. Of course, this depends on the type of technology used for both the LMUs and LIs. On the other hand, if we offer only real-time data services, the number of LMDS Infostations required for continuous service could decrease due to the larger permissible delay separation between adjacent base stations. This is true because of the less stringent delay requirements for data applications.

As aforementioned, LMDS Infostation was just one approach to offering train commuters continuous services. The following section will discuss another solution known as LMDS Tower Sites.

4.1.2 LMDS Tower Sites (LTS)

This section discusses the potential use of another approach known as LMDS Tower Sites (LTSs). We examine the economical feasibility and the link budget requirements.

4.1.2.1 Architecture

LTSs have two transceiver and antenna units on opposite ends of the tower. These units will alternately take control, depending on whether the train is departing or approaching an LTS. Figure 4.7 provides a visual illustration of this architecture.

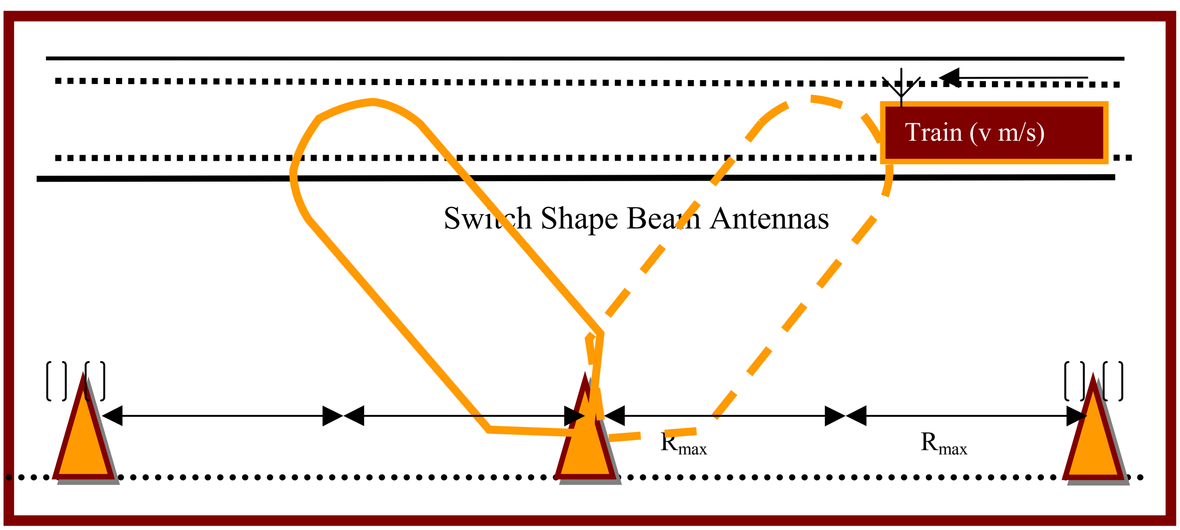

Figure 4.7: LMDS Tower Site Architecture

Figure 4.7 provides an illustration of the potential system layouts for our LMDS Tower Sites. This figure shows the antenna coverage area of adjacent LTSs. A switching method would be used to control the operation of the two antennas during the on-axis crossover period of the two antenna beams. This is needed to avoid the waste of power by having two transceiver and antenna units operating simultaneously. If the antennas were operating at the same center frequency, then the gain of the antennas would just combine at the crossover point and result in a higher transmitted signal to the receiver.

Another alternative method with this same paradigm involves the frequency channel selection. Should the system be designed to use different sets of frequency channels at one tower site? With this approach, the tower sites would just be pre-configured to operate with different frequency pairs. This could increase the complexity of the receiver unit because it would have to be reconfigured to operate at different carrier frequencies quickly. We would still be required to control the required switching at the LTSs. Another interesting fact is the number of needed mobile units on the train is the same as the LMDS Infostation approach, ONE.

We next examine the distance required between each tower site. Unlike the LMDS Infostations approach, the maximum distance separation will be determined by the systems capability to meet the required gain of the transmitting antenna. We will further elaborate on this requirement in the following section. Table 4.4 evaluates the relationship between the coverage distance, the required transmitter gain, and the number of LMDS Tower Sites needed to supply continuous coverage. The total coverage area, C_A, is determined by $2*R_{max.}$ The number of LMDS Tower Sites is determined by (Track Length/C_A). A constant value for path length, 230 miles, is used.

Table 4.4: Number of LMDS Tower Sites

Coverage Area		**Number of Required Stations**
R_{max},	C_A	
2km	4km	93
3km	6km	62
4km	8km	47
5km	10km	37

4.1.2.2 Link Budget Analysis

The LMDS Tower Sites (LTS) have different link budget parameters because of the new requirements on coverage area. Table 4.5 gives a set of new link budget parameters suitable for this approach.

Table 4.5: Link Budget for LMDS Tower Sites

Parameters	**Intermediate Parameters**	**Values**
P_t	500mW	27 dBm = -3 dBW
$\mathbf{G_t}$ (Shaped Beam)	*Will vary depending on desired coverage area.*	35 dB
$\mathbf{G_{AR}}$ (Pyramidal Narrow Beam Horn)	H-θ_{3dB} = 26° E-θ_{3dB} = 23°	16.2 dB
$\mathbf{L_p}$	R = varies between 5.7m and transmitting distance (ex. 2km)	Using worse case (2km), we get 128 dB
$\mathbf{L_{EX}}$		14 dB
$\mathbf{T_s}$	900K	30 dB
K (constant**)**	$1.38x10^{-23}$ J/K	-228.6 dB
$\mathbf{R_b}$	300Mbps	84.8 dB
$\mathbf{B_n}$	150MHz	81.8 dB
$\mathbf{I_m}$		2 dB
C/N calculated		**18 dB**
$\mathbf{E_b/N_o}$ **calculated**		**21 dB**
$\mathbf{E_s/N_o}$ **required**	4 FSK	**13.8 dB**
$\mathbf{E_b/N_o}$ **required**	4 FSK	**10.8 dB**
$\mathbf{E_b/N_o}$ **margin**		**21 – 10.8 = 10.2 dB**

The parameters from Table 4.5 were selected to meet the specifications in Chapter 3. The LTS approach introduced issues, such as additional losses, which required tradeoffs between the transmitted power and the gain of the transmitting antenna in order to have a sufficient E_b/N_o link margin. Explanations for these changes are below:

- $\mathbf{P_t}$ was increased due to the additional requirement on coverage area. In order to compensate for the large coverage area, we must consider a tradeoff between the transmitted power and G_t. The other link budget parameters are fixed.

- $\mathbf{G_t}$ was approximated by considering the requirements of using shaped-beam antennas. The selection of transmitting antenna was a very essential part of this design. The antenna must be designed to transmit a large amount of information over a short direct path. We would design our antenna to have a broad horizontal pattern, and the vertical pattern would vary depending on the height of the tower site and the positioning of the antenna relative to the mobile unit.

- $\mathbf{G_{AR}}$ was derived using Eqs. 4.1 – 4.3. A large narrow-beam antenna was selected because we desire to have a concentrated directional beam while communicating with the small shaped transmitting beam.

- $\mathbf{L_p}$ was determined by considering the maximum transmitting distance. The path length from the transmitter (TX) to the receiver (RX) now varies between the right-of-way distance (5.7m) to (R_{max} = 2000m). R_{max} varies depending on our desired coverage area.

- $\mathbf{L_{EX}}$ was doubled from the LMDS Infostation approach because of the additional distance between the TX and the RX. Thus, the greater the distance, the larger the excess path loss.

4.1.2.3 Summary

We can conclude that the LMDS Tower Site approach requires a smaller number of units. This is largely because of the extended coverage areas when using shaped-beam antennas. Of course, the design tradeoff is the larger transmitter gain requirement, which in turn increases the power requirement. One disadvantage could be the system cost involved with two transmitter, antenna units at one tower site. Nevertheless, this approach does have potential advantages in terms of data routing and robustness in a dual-train environment. These aspects and others relative to the LMDS Tower Site approach will be further discussed in section 4.2.

4.1.3 LMDS Tower Site/Infostation (LTS/I)

The third methodology takes advantage of simulcasting as its transmission technique. Simulcasting is widely used today in Personal Communication Systems (PCS), such as paging. Simulcasting allows the same information to be broadcast simultaneously over a pre-determined region. As a comparison, we will assume that the coverage region is the same as that of Figure 4.7 (i.e. R_{max}). It must be noted that this approach is actually a hybrid solution between the LMDS Tower Site and the LMDS Infostation approach. We would use broad fan beam antennas at the transmitting base stations, unlike the LMDS Infostation (LI) approach, which used these antennas at the receiver end. In this scenario, the fan beam would direct most of the energy in the vertical direction towards the train (see Figure 4.2 and Figure 4.8). Of course, the coverage average is determined by Eq. 4.6, which is a factor of the fanned beamwidth.

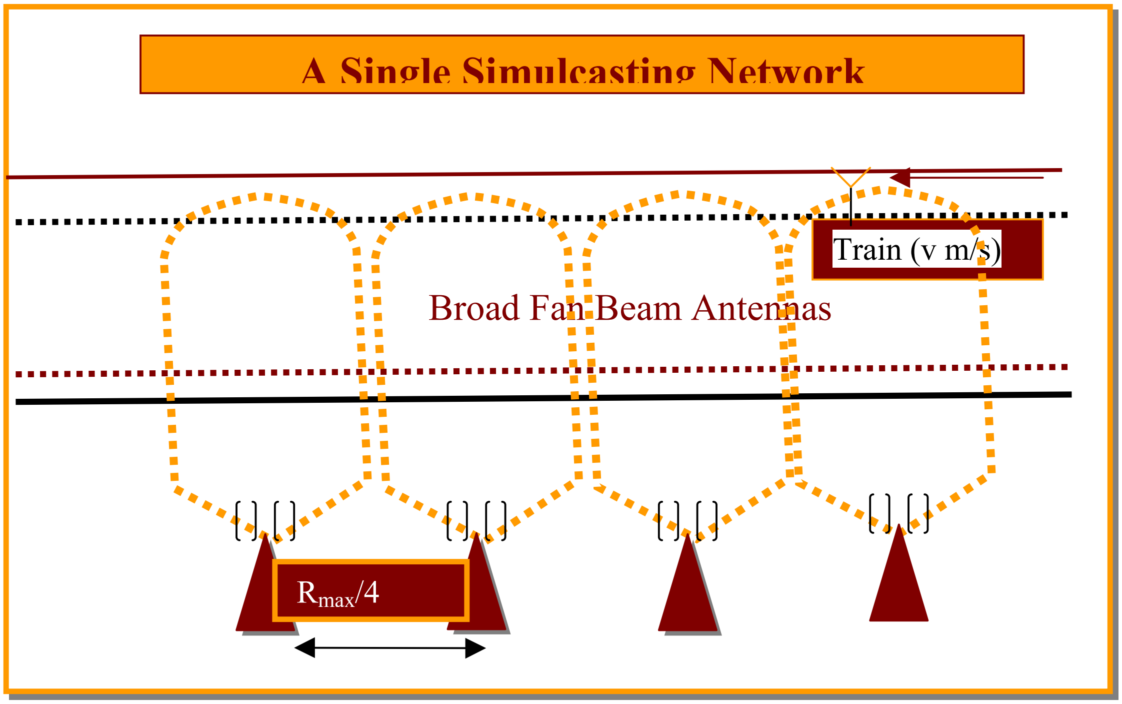

Figure 4.8: Simulcasting LMDS Tower Site/Infostation Architecture

The LMDS mobile unit would communicate with the base station in which it receives the highest power reception. This process would avoid the use of a switching mechanism as with the LTSs because the units would no longer be co-located. In addition, our distance range would be smaller, which would limit the amount of power needed at each site. We would assume for every proposed LMDS Tower Site (LTS) there would be two low-powered LTS/I sites located one-half the distance of any LTS. Thus, if the total distance, R_{max}, was 4km, we would need four LTS/I units operating at presumably one-fourth of the required power of the LTS unit. We would still need to insure that the simplex channels for transmitting and receiving have enough frequency spacing separation in the spectrum to avoid interference. This also implies that we could use a multicasting mode at the terrestrial network level to broadcast the same information to the respective LTS/I units within a certain region. This process could be advantageous in terms of the decrease in data routing and handover required.

One disadvantage of this approach would be the increase in needed base stations. However, if the cost of one LTS is at least four times greater than the cost of a single LTS/I, then this method would become more economical with a potential added improvement in terms of data routing at the terrestrial network level. Currently, an industry price of LMDS transmitters and receiving units requiring 50mW or less ranges from $5K to $10K. However, these units could potentially be made in-house at cheaper prices.

We have described three approaches to offer continuous services to train commuters. The following section will serve as a summary by providing an in-depth discussion for the best methodology.

4.2 Making the Best Selection

In selecting the best architecture, a list of requirements was developed. A system must have the following specifications:

- Meets the system throughput requirements within a given set of constraints
- Economical feasibility (i.e. reasonable number of base stations requirement)
- Minimizes system complexity
- Suitability for serving multiple trains
- Minimizes networking and data routing requirements

Before discussing the most suitable approach, we will revisit and compare the three methodologies (see Table 4.6).

Table 4.6: Comparison of Three Methodologies

Approach *Attributes*	**Infostations**	**Tower Site**	**Infostation/Tower Site**
Power Requirement	Very Low (ex. 1mW)	High (ex. 500mW)	Between Infostations and Tower Site power range
Antenna Requirements	• Broad beam (fan) at receiver • Narrow beam (horn) at transmitter	Two transmitter-antenna units at each site	• Broad beam at transmitter • Narrow beam at receiver

Table 4.6: Comparison of Three Methodologies

Approach *Attributes*	**Infostations**	**Tower Site**	**Infostation/Tower Site**
Data Transmissions	Uses Simulcasting approach. Information is transmitted in constant short burst.	Continuous transmission	Simulcasting approach. Because of broad beam at transmitter we have longer transmissions than the Infostations approach.
Number of Required Units (230 miles coverage area)	Very large (ex. 3,000) depending on the beam width of the transmitting antenna and speed of the train.	Very small depending (ex. 93) on the required coverage area.	Approximately four * the number of Tower Site units.

In critically examining Table 4.6 and the previously mentioned material, the LMDS Tower Site (LTS) approach meets the criteria with fewer disadvantages as compared to the other approaches.

Table 4.7 provides advantages and disadvantages for the LTS approach.

Table 4.7: LMDS Tower Site Approach Evaluation

Criteria	**Advantages**	**Disadvantages**
System Throughput Capability	Can meet the required throughput at any reasonable speed. The throughput capability is shown in *Figure 4.9*.	
Economical Feasibility Regarding Number of Base Stations	Requires fewer base stations than other approaches. *See Table 4.7.*	
System Complexity	Offers a continuous stream of data flow over a greater period. Advantage in maintaining synchronization with mobile unit (s).	✗ Requires two transmitter and antenna units per Tower Site. Could increase power requirement. ✗ Could involve switching or a timing mechanism to control units.
Suitability for Serving Multiple Trains	Can alternate the roles of units located at each Tower Site (i.e. Both independently serving two trains).One beam would cover the same territory as two beams were covering (R_{max})	

Table 4.7: LMDS Tower Site Approach Evaluation

Criteria	**Advantages**	**Disadvantages**
Networking and Data Complexity	Data routing is inherently minimized due to the decrease in number of required base stations. Single buffer used to serve both units at one tower site.	As with any approach, must determine the appropriate site for data routing. We could have a Tower Site Control (TSC), which routes the data to several Tower Sites within a pre-defined area. Another option would be to have one of the LTSs function as the control center for a certain region. (see Figure 4.10)

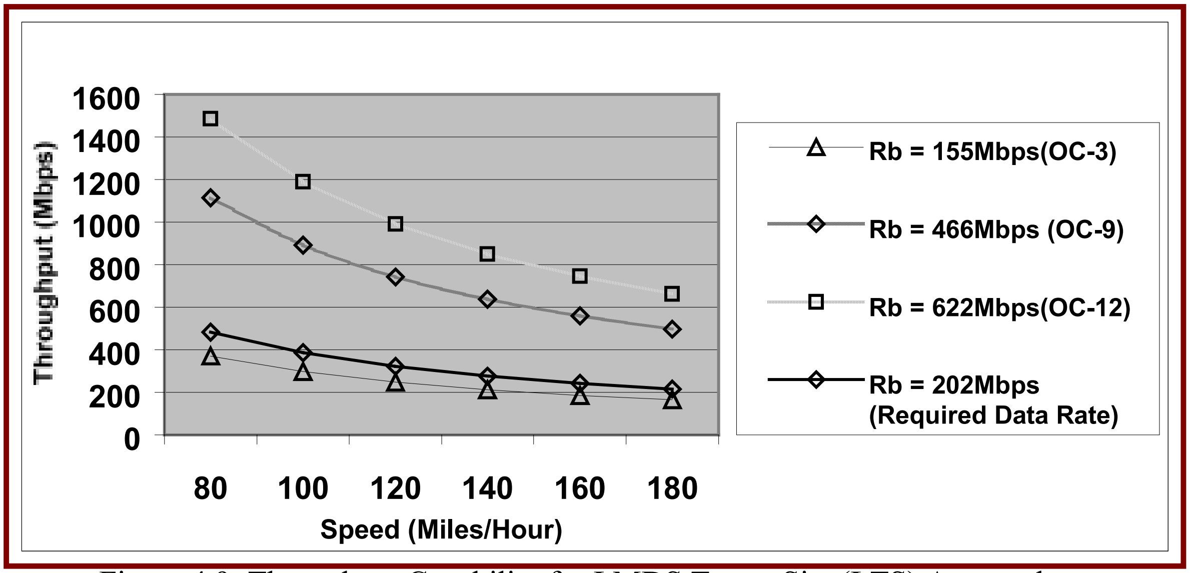

Figure 4.9: Throughput Capability for LMDS Tower Site (LTS) Approach

Table 4.8: Economical Feasibility Regarding Required Base Stations

Required Number of Bases Stations given A 230 miles track length	
LMDS Infostations	3066 (given a speed of 120mph) **See Figure 4.6**
LMDS Tower Sites	93 **See Table 4.4**
LMDS Infostations/Tower Sites	93*4 = 372 (given 4 stations/LTS) **See Section 4.1.3**

Having explored the essential reasons for selecting our system architecture, we will briefly discuss potential networking solutions.

4.3 Potential Networking Solutions

4.3.1 Transport Technique

Currently, industry is debating whether ATM or IP provides the best common infrastructure. This concern must be taken into consideration when selecting the optimal packet format for our data transmission. Although our focus transport channel is wireless, we still wish to select a format that is adaptable to the other two networks (i.e. terrestrial network and the network on the train). Table 4.9 gives a brief comparison of the advantages of ATM versus IP.

Table 4.9: ATM versus IP Attributes [21]

Attributes	**ATM**	**IP**
Capable of multiplexing services (video, data, and telephony)	Already Exist	Working towards a Multi-service IP Carrier Network. [23]
Meeting QoS	Yes	Working towards this service.
Traffic speeds conform to standard telephony transport rates (e.g. DS-3 (44.7Mbps) or STS-1 (51.8 Mbps))	Yes	Maybe in the future
Industry expertise in technology	Newer technology, not as much known knowledge as IP.	Yes
Current Existing Networks	Networks exist and are currently under structure.	IP networks exist (data transport)
PC software and drivers for on train network.	Yes	Yes

Currently, there exists an effort to migrate to a multi-server IP Carrier network. This network's aim is to provide those same QoS metrics as provided by ATM. IP is primarily known to support data applications, however, since a majority of the networks carry IP traffic, work is being implemented to study the potential of using IP to transport voice, video, and data simultaneously. As mentioned earlier, IP attributes that are primarily associated with the use of datagrams are issues that must be resolved if IP is to be used as the next common infrastructure.

Although ATM was originally designed to work in fixed wired high-speed environments, it has been proposed and tested to work in wireless arenas as well. Since IP is one of the most popular transports for data, another possible solution is to use an IP/ATM paradigm. This approach would potentially lend itself to meeting QoS requirements of all traffic types. (i.e. voice, video, and data) This issue is further discussed in the following section on mobility management.

4.3.2 Mobility Management

This section will discuss mobility management attributes required for a Mobile Application Part (MAP). We will not discuss major details of protocols currently being studied, for this is beyond the scope of the thesis. The goal of the MAP would be to allow seamless transport of data among LMDS Tower Sites (LTS) or LTS/I networks. The functions of the MAP would include authentication of the mobile unit, transferring data to the visiting network, and mechanisms for routing connections between LTS units. A visited network is a network that is not the subscriber's original operating network. These functions would be considered as part of the mobility management protocol in our mobile architecture. Two current common air interfaces, Global System for Mobile Telecommunications (GSM) and IS-41, have defined a MAP. IS-41 allows different service providers of Mobile Switching Centers (MSCs) to pass information about their users to other MSCs on demand. This is a similar paradigm to our network because in reality the train will be passing through distributed independent LTSs throughout the train route. This MAP would also define the application protocols between switches and databases for supporting call management, location management, security management, radio resource management, and mobile equipment management. [24]

The MAP can become quite cumbersome. To reduce complexity, we could possibly just implement Mobile IP and/or Mobile ATM. Mobile IP allows a user who is attached to a home network to visit other IP networks while remaining attached to his original network. The communication is implemented by transmitting IP packets to the home agent and then encapsulating and sending them to the mobile node via a foreign agent. A foreign agent allows the mobile node to communicate with the Internet while it is away from the home network. A foreign agent is a router that assigns a temporary worker care-of-address to the mobile user for bi-directional communication. It also has the task of cooperating with the mobile node to ensure smooth handoffs, if needed. This is done by being careful not to drop the datagrams even when the mobile node has moved away from the care-of-address receiving the datagrams. [25] See reference 25 for further research interest in Mobile IP.

Work is also being done in the area of Mobile ATM. Mobile ATM covers similar aspects of both Mobile IP and MAP. It addresses ATM signaling extensions for handoff control, location management, and mobile-QoS control. Mobile ATM framework focuses on offering real-time streams with QoS in both wired and wireless networks. This will be desired in our system architecture because of the connection transport needed for interoperability among all three networks in our system architecture. Work is also being studied for the potential use of IP-over-mobile WATM and/or Mobile IP over WATM. For a more in-depth discussion on this area, consult reference [26].

4.4 A Glance at the On-Train Network Design

As the data is continuously transmitted to the receiving antenna in our LMDS channel, it must be routed to the appropriate user on the train. As alluded to earlier, each user desiring to use the services could be connected to a wired Gigabit Ethernet, Fast Ethernet or ATM switch. The network infrastructure must be designed to have asymmetric communication between the wireless LMDS network and the wired on-board network. We can accomplish this by connecting an Ethernet switch box to the mobile LMDS transceiver unit on the train. This Ethernet switch would then connect to both an AC unit and other network components. Each Ethernet port could have the capability of allowing the user to connect to either a 10/100Mbps or 1000Mbps RJ-45 connection located in the seatback. There might also be a need to have cable connections to routers, hubs, servers, and other switches. Finally, a Network Interface Card (NIC) would be used to allow the connection between the PC and the Ethernet switch. Two popular vendors of Ethernet equipment are Cisco and Lantronix. This work would require further research of the optimum and/or appropriate network components for our on-board infrastructure.

4.5 Summary of Approaches

This thesis has served to provide an overview of the potential design concerns associated with the deployment of a high-speed mobile LMDS system. We have discussed three approaches for offering continuous services to our train commuters. A detailed system evaluation was provided for the best methodology, LMDS Tower Site (LTS). Figure 4.10 (see following page) gives an overall illustration of the LTS approach. It contains important issues discussed in Chapter 2 – Chapter 4.

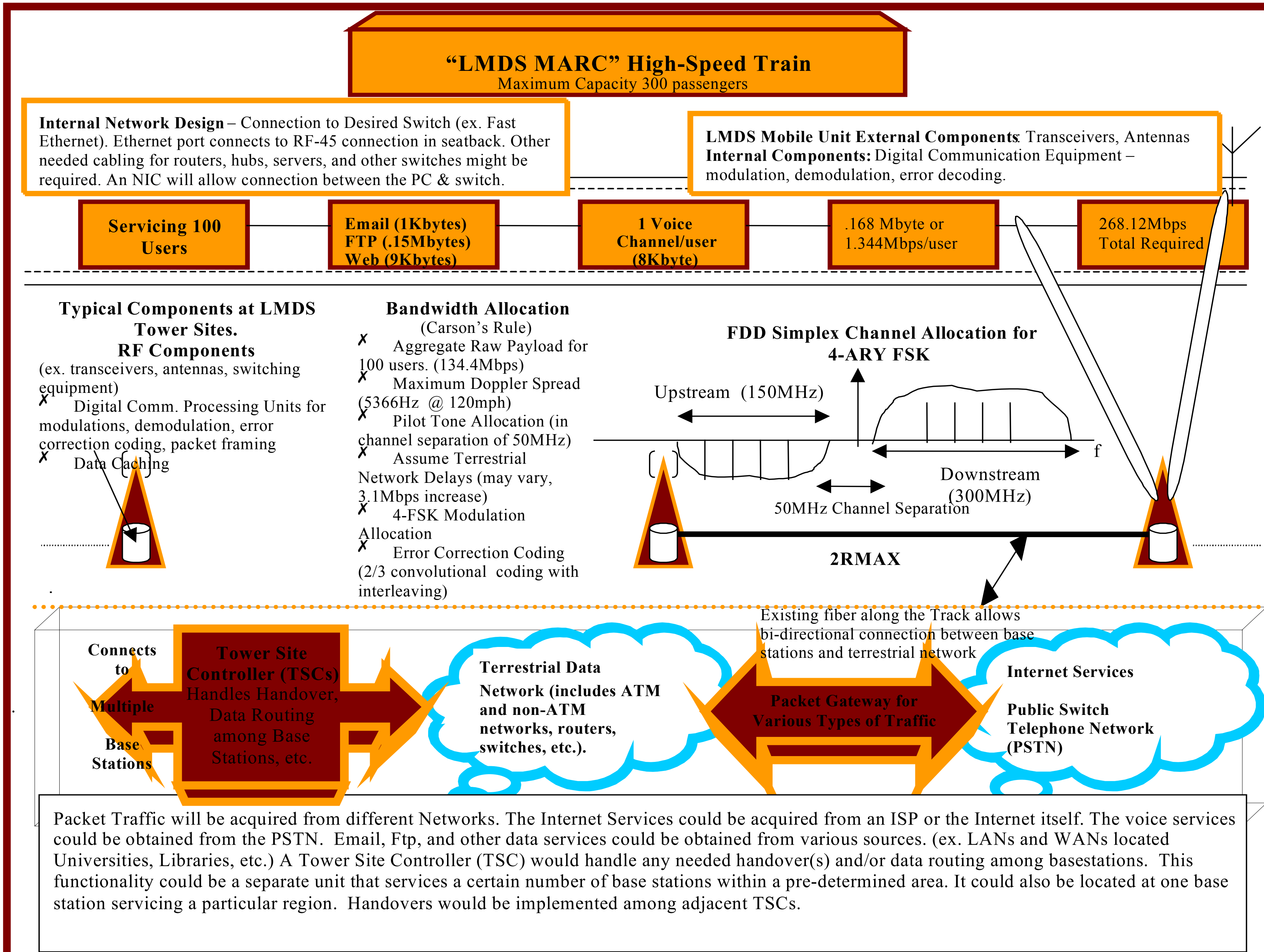

Figure 4.10: LMDS Tower System Design for A High-Speed Mobile Train

Chapter 5

Conclusion and Suggested Future Research

This thesis provided a broad overview of three system architectures: LMDS Infostations, LMDS Tower Sites, and LMDS Infostations/Tower Sites. The system architectures focused on offering continuous broadband services in a high-data rate mobile LMDS environment. Modulation selection, duplexing, multiplexing, error correction coding, and other key physical layer issues were discussed relative to our high-speed mobile environment.

We chose N-FSK as a suitable modulation technique because of its lack of phase sensitivity. We can transmit a huge amount of information in our high-speed environment using FDD as the duplexing technique. We selected ATM because of its ability to multiplex voice, video, and data seamlessly and offer QoS for various applications. We selected an FEC technique, 2/3 convolutional code because it allows continuous error detection and correction in our robust environment. We also used interleaving to guard against channel burst errors.

The system was designed to meet a cumulative data rate of 134.4Mbps. We estimated an overhead amount of 597ms for data routing. The Doppler shift experienced at 120mph was 4878Hz. The estimated required bandwidth was 263MHz, when we considered channel coding, routing overhead, and the Doppler effect.

We also examined link budget parameters, such as power levels, antenna gains, and noise bandwidths. A low power level of -30dBW could be used in the Infostation approach; however, a high-transmitted power of -3dBW was required with the LMDS Tower Site (LTS) approach because the transmission distance increased. Because of the distance, we also increased the antenna transmission gain from 20dB to 35dB, respectively. The E_b/N_o link margin for the LTS approach was only 10.2 dB as compared to 42.4 dB with the Infostation approach.

In selecting the best architecture, we used a set of pre-defined criteria, which included throughput capability, economical feasibility, system complexity, data routing, and robustness in serving multiple trains -- all of which encompass the previously discussed information. We selected the LMDS Tower Site approach as the best design primarily because of its suitability to serve multiple trains, economical feasibility, and decreased traffic load during data routing. This approach can potentially take advantage of existing infrastructure, thus reducing the overall system design cost.

As with any system architecture, simulations and/or modeling would provide a warning of potential bottlenecks and/or tradeoffs before actual development. There are many opportunities for further research within this architecture. One particular area is the behavior of other candidate modulation techniques in our robust environment. Another exploration area is the implementation of meeting QoS for various applications in our high-speed environment. This research could focus on examining the potential use of IP, ATM, or a hybrid solution, IP/ATM during channel transmission. Other research could focus on investigating critical networking components needed to provide seamless interconnection between the LMDS Tower Site and the terrestrial links. This new research may offer further insight to other system architectures for our high-speed mobile LMDS system.

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Vita

Katina R. Reece

Katina R. Reece was born on July 21, 1974 in Macon, MS. She earned her B.S. degree in Computer Engineering at Mississippi State University (MSU) in December 1997. While attending MSU, she actively participated in several organizations, such as IEEE, Eta Kappa Nu, Alpha Kappa Alpha, and Increasing Minority Accessment to Graduate Education (I.M.A.G.E.). During her undergraduate tenure, she completed three cooperative education sessions with the National Security Agency (NSA) and one internship with Exxon Company, USA. She received several cash awards during her cooperative education program. Katina also received honor recognitions, such as "Highest Minority GPA in College of Engineering Award" for two consecutive years and the National Physical Science Consortium (NPSC) fellowship. The NPSC fellowship is a 6-year, $200,000 program between a sponsoring company and any Ph.D. granting institute in the U.S. Her sponsoring company was NSA and her chosen university was Virginia Tech. In 1998, Katina joined the Center for Wireless Telecommunications (CWT) at the Bradley Department of Electrical and Computer Engineering. She participated in many Local Multipoint Distribution Services (LMDS) research efforts. She obtained her Masters of Science in Electrical Engineering in December 1999. Her current research interests include system design, wireless networking, and network modeling and performance analysis.