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A Realistic, Survivable Packet Radio Network Design for Mobile Multimedia Communications

by

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Abstract

The provision of multimedia services to a wireless environment presents several technical challenges. The extension of these services to a survivable, distributed system adds a significant level of complexity. Packet radio network technology offers one possible starting point for a solution. This thesis approaches the problem of delivering integrated voice, low grade video and low rate data services to a wireless network operating in a distributed environment through the use of a packet radio architecture.

An overview of the technical specifications to be addressed in the network design is used to define the operational requirement. A radio channel model based on the Hata-Okumara approach is then developed in order to provide a realistic simulation framework for the system and a basis for network connectivity maps in subsequent chapters. The issues related to the operation of a distributed network are then explored, essentially illustrating the design implications of survivability. Several different multiple access techniques are also evaluated for suitability in this network design. Based upon the criteria provided at the onset, a slotted CDMA methodology is recommended. This leads to the establishment of network control parameters including the data structure, routing and scheduling schemes based on slotted CDMA operation in a distributed environment. Finally, issues related to network integration and enhancements are discussed, including CDMA code orientation, control channel modifications and acknowledgements. Recommendations for future work are also provided.

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(in order of use)

α	alpha – used to represent the first baseline data set
β	beta – used to represent the second baseline data set
γ	gamma - used to represent the third baseline data set
δ	delta - used to represent the fourth baseline data set
ε	epsilon - used to represent the fifth baseline data set
σ	standard deviation
P_r	power received
P_t	power transmitted
G_t	antenna transmit gain
G_r	antenna receive gain
h_b	base station antenna height
h_m	mobile station antenna height
L_p	path loss
f_c	carrier center frequency
$a(h_m)$	adjustment based upon mobile antenna height
R	range
No	noise floor
k	Boltzman's constant = 1.38×10^{-23} Ws/k
T	system noise temperature
B	bandwidth
M	the maximum number of simultaneous transmissions
E_b/N_o	the power to noise ratio dictated by the bit error rate
R_b	data rate
R_c	chip rate
A	number of pair links
N	the number of nodes
$D_{ij}^{(h)}$	the optimum path from a source node i to a destination node j in a network with link length h
d_{ij}	is the 'cost' of any point to point link between transmitting node i and receiving/target node j

1.0 Introduction

1.1 Background

With the promise of "anytime, anywhere" communications, wireless networks have attracted significant attention in recent years. The application of these systems to mobile scenarios or to more static services in austere environments is achievable today and has been for some time. However, with the added requirement for rapidly deployable systems capable of operating in multi-hop environments and no fixed infrastructure, the ability to provide multimedia connectivity becomes much more constrained and complex. The resulting void has spawned considerable additional research into so-called third generation systems, although this has yet to provide a clear way ahead.

Wireless multimedia has many possible applications. In particular, the military has demonstrated significant interest both in Canada and abroad with a specific view of extending multimedia information to the battlefield. Indeed, a major US Department of Defense initiative has the aim of extending systems from the "White House to the foxhole".

For the soldier in a trench it is not hard to understand the intrinsic value of a detailed map complete with intelligence overlays, digital imagery or weather information. Of even greater significance are the capabilities these technologies could bring to commanders at all levels if this information can be shared, merged and manipulated over a wide area. The ability to craft a common operational picture is widely acknowledged as a force multiplier of such extreme importance that numerous heavily funded projects are underway.

Although military interest is driving some of the research in this realm, it should be noted that applications are not limited to the military milieu. Clearly, there is now a demand for such systems in the fields of law enforcement, fleet management, disaster recovery, support to special events (e.g. conferences) and Search and Rescue. It is possible that demand for commercial goods based on this technology is about to skyrocket.

Unfortunately, while efforts have been widespread, the band-limited nature of the wireless environment has presented numerous challenges that have yet to be overcome to adequately meet the demands of these applications. One concrete illustration of the challenges that remain can be observed in the Call for Papers in recent issues of the *IEEE Journal of Selected Areas in Communication.* Relevant subjects include: Broadband Wireless Techniques (June 98), Spread Spectrum for Global Communications (April 98), Wireless Ad Hoc Networks (January 98) and Broadband Wireless Techniques (June 97).

Packet radio networks offer one possible architecture through which multimedia information could be relayed in the wireless environment. Indeed multimedia requirements appear to have revived interest in packet radio network design.

Leiner et al's seminal paper [1] on the subject of packet radio network design raises the principal relevant issues that must be considered in undertaking such an effort. Specifically, they concentrate on issues revolving around physical connectivity, channel access, link determination, routing, congestion and flow control. The topics are in turn inherently linked to many other factors, for example, requirements related to the data class being passed, and the environment in which the system is fielded.

The topics in [1] have been used as a starting point for this multimedia packet radio design and are modified to form the framework of this research. A general network design will be conceived to provide an architecture capable of supporting the capabilities and requirements outlined below. It will become evident that such a network design involves a series of trade-offs between control, connectivity, capabilities and quality of service (QoS) in which it is impossible to achieve an optimum solution for all situations. However, it is possible to describe a design that provides solid performance under specified constraints.

1.2 System Requirements

Before beginning the actual design of any system, it is first necessary to define the parameters that the system must support. In military terms this is referred to as definition of the 'operational requirement'. These requirements and their resulting implications will have a direct impact on the feasibility of the various options available to the designer.

1.2.1 Integrated Multimedia Wireless/Mobile Operation

First and foremost, the system to be designed here will employ wireless packet radio technology to provide rapidly deployable, mobile multimedia connectivity. The multimedia capabilities to be provided will consist of voice, video and data traffic integrated on the common radio platform, at the rates defined below. Given that RF technology is to be employed, the development of an appropriate channel model becomes necessary.

The seamless integration of multimedia traffic is in itself an interesting problem with its own considerable body of research. It implies an inherent capability to provide both an adequate signal to noise ratio to support reliable transmission for each data type and a suitable Quality of Service (QoS) guarantee for the delay sensitive traffic.

1.2.2 Survivability and Distributed Control

Secondly, the decision is made that the overall system must be survivable. This in turn implies a distributed architecture in which there are no base stations analogous to cellular base stations and, in fact, there is no fixed infrastructure whatsoever. All nodes, terminals or stations (the terms are used interchangeably in this research) are identical in terms of hardware, but some may be selected, based upon their location and connectivity, to be key stations. This assumption adds the requirement for additional overhead as no central station will be omniscient or omnipotent. Individual stations will need to be aware of the network's topology to allow for decisions to be made with respect to scheduling and routing.

Survivability has many advantages and is particularly useful in the case of military units on the battlefield where stations may "crash" for less benign reasons than in the civilian case. However this characteristic also adds challenges of its own to the system designer.

The decision for the network to be 'survivable' is an absolutely critical one, particularly when combined with the previous requirement to provide integrated multimedia support. The vast majority of research in the wireless field has been applied to conventional situations with base stations and centralized control. The removal of such stations introduces unique complications requiring unique solutions and trade-offs. In a true distributed network, the addition of the requirement for QoS guarantees can prove to be especially challenging in view of the fact that QoS guarantees further imply the reservation and control of resources.

1.2.3 Users and Range

One final arbitrary design assumption that is made is for the system to support 25 terminals spread over a 20 km by 20 km grid of terrain. Again, this is not a purely military consideration, but it is easily applied to a small military unit of platoon or company strength occupying territory or to law enforcement bodies operating in a set piece of land.

1.2.4 Initial Data Estimates

The users in the network must have the ability to transmit voice, data and low grade video traffic. One example from Mermelstein et al [2] defines the momentary requirement for a 500 terminal organization is provided below in Table 1.1.

Type of call	# of calls	Avg Tx Rate	Total Rate Reqd
Data (low speed)	20	2.4 kbps	48 kbps
Data (mid speed)	20	9.6 kbps	192 kbps
Voice	80	5 kbps	400 kbps
Video	5	32 kbps	160 kbps
Total			800 kbps

Table 1.1: Data rate requirement – 500 terminal network [2]

In this example, an 800 kbps aggregate data rate would be required to support the average throughput. However, these statistics are based on a system with presumably a much larger demand. Of greater import is that this aggregate must be based upon a network in which nodes are no more than one hop apart, otherwise, while the effective data rate would be 800 kbps, the actual data rate would be much higher.

In our case, instead of supporting 500 terminals, we are exploring a 25 station network and therefore the data transport requirement, in general, would be considerably lower. Consequently, there may be some flexibility to increase the average speed proposed for video. Table 1.2 contains the extrapolation of the data from Table 1.1 for a 25 station network.

Type of call	# of simultaneous calls	Average Tx Rate	Total Rate Required
Data (low speed)	2	2.4 kbps	4.8 kbps
Data (mid speed)	2	9.6 kbps	19.2 kbps
Voice	4	5 kbps	20 kbps
Video	3	32 kbps	96 kbps
Total			140 kbps

Table 1.2: Data rate requirement extrapolated for a 25 user network

Table 1.2 is not a linear extrapolation of Table 1.1 as it is modified for higher usage rates. For a small military unit engaging, for example in combat or peacekeeping operations, it is likely that the demand in general would be higher, and in some cases almost continuous. Using this model, an estimated total data rate of 140 kbps would be required. It must be noted that this calculation does not include the requirement to provide a control channel that is required to allow individual terminals to determine the network's topology and overall connectivity, amongst other functions. This will be added.

In view of the aim to provide integrated multimedia services, some simplification can be made at this point to provide for scalability. Specifically, if 2 kbps is used as a baseline building block for the various services, it is likely that integration of these data types would be simplified. Therefore, the following is proposed in Table 1.3.

Type of call	**# of simultaneous calls**	**Rate**	**Effective Data Rate Required**
Low Speed Data (LSD)	2	2 kbps	4 kbps
Medium Speed Data (MSD)	2	8 kbps	16 kbps
Voice	4	4 kbps	16 kbps
Video	3	32 kbps	96 kbps
Control	1	16 kbps	16 kbps
Total			148 kbps

Table 1.3: First modification - network data rate requirements

The data rates selected for voice, video and control require further explanation. Great strides have been made in the areas of data coding, thereby permitting the transmission of voice and video at lower data rates than previously thought possible. The development of advanced compression techniques has contributed significantly to this trend.

Voice has traditionally been digitized at 64 kbps (the 4 kHz voice channel has been sampled at 8000 samples/sec and quantized by 8 bits per sample). Presently, the G.729 ITU standard utilizes the conjugate structure algebraic code excited linear predictive (CS-ACELP) algorithm for compression of voice from 64 kbps to 8 kbps. G.729 is the default Frame Relay Forum voice coder. It is expected that further advances will be achieved in the short term and therefore 4 kbps voice is an acceptable assumption for use in this analysis (at present rates it would be acceptable if the voice coder sampled at a lower frequency or digitized to fewer levels).

The video estimate is based upon a requirement for low-grade video, whiteboard and pre-stored maps with overlays. The MPEG-4 standard has been developed to address the requirement for video in bandwidth limited environments such as wireless. MPEG-4 is an ISO/IEC standard (ISO/IEC 14496) developed by the Moving Pictures Experts Group. This will become an international standard in 1999 and will provide a very low bit-rate video (VLBV) core for applications operating at 64 kbps and less. This will support image sequences with low spatial resolution and low frame rates. Thus, 32 kbps is a reasonable assumption for low-grade video.

The initial control channel estimate of 16 kbps was based on the requirement for each node to transmit enough data to indicate its own connectivity and that of its neighbors. At this point 640 bps of control channel capacity per node is thought to be sufficient for a basic control capability. This would allow for binary transmission of the node ID number, the node's connectivity table (a matrix of 625 bits) and several flags as required. The control channel will be examined in much greater detail and is fully defined in Chapter 5.

1.2.5 Data Rates and Virtual Circuits

Table 1.3 indicates that the baseline requirement is to provide 148 kbps of throughput in the system. However, key elements are still missing from this estimate regarding virtual circuit provision for delay sensitive voice and video links.

In order to achieve an effective data rate of 148 kbps, the system designed must be capable of higher actual data rates if Virtual Circuits are to be in place and connections are maintained between nodes that are more than one hop apart. This is because a terminal cannot simultaneously transmit and receive the same channel, as illustrated in Figure 1.1 below, using a half duplex and full duplex video examples for one hop and greater than one hop examples.

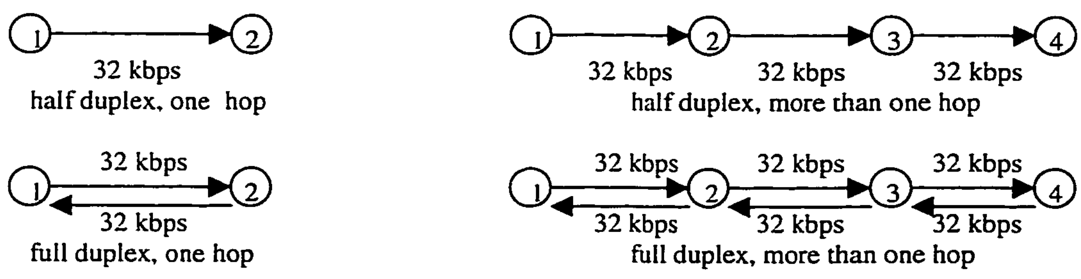


Figure 1.1: Data rates for VCs – half and full duplex, one hop and more

In Figure 1.1, we observe a major difference between one-hop and multi-hop links. A simple one-hop, half-duplex link for video at the rates specified above will require 32 kbps. If this is upgraded to a full duplex link, 64 kbps is required. However, much more is required in the multi-hop case if simultaneous transmissions are not possible.

In this example, intermediate nodes (here nodes 2 and 3) on a multi-hop, half-duplex link will require 64 kbps: 32 kbps in and 32 kbps out if a constant data stream is to be maintained. The originating node and final destination node only require 32 kbps each. For a system where only one station can transmit at a time, such as a TDMA network in a small area, the actual data rate required to support the circuit would be three times 32 kbps, or 96 kbps. If this circuit is then upgraded to full-duplex operation, then the requirement at each intermediate node becomes 128 kbps (32 kbps in from each direction and 32 kbps out from each direction). The overall throughput required to support a two-hop, 32 kbps full-duplex link is then 192 kbps.

It is assumed that some multi-hop links will have to be maintained, although to a large degree the final estimate of the system's data rate will be dependent upon the multiple access technology that is employed. Therefore, further discussion of the system's data rates will take place in Chapters 5 and 6 after the multiple access technique has been selected. However, at this point, to minimize the overall effect of the half-duplex nature of the voice and video circuits, another modification to the systems data rates is deemed appropriate.

Type of call	**# of simultaneous calls**	**Rate**	**Effective Data Rate Required**
Low Speed Data (LSD)	2	2 kbps	4 kbps
Medium Speed Data (MSD)	2	8 kbps	16 kbps
Voice	4	8 kbps	32 kbps
Video	3	32 kbps	96 kbps
Control	1	16 kbps	16 kbps
Total			164 kbps

Table 1.4: Second modification - network data rate requirements

The modification shown in Table 1.4 consists of two changes. First, the data rate allocated for voice use is doubled to accommodate its half-duplex nature (i.e. 4 kbps per direction). Second, the actual data rate for the video traffic is halved to 16 kbps, and then doubled, again to accommodate half-duplex links. It is acknowledged that this will severely impair the ability to carry true video. However, such a link would still be appropriate for very low grade video (it is only ¼ the rate of the peak MPEG-4 rate) and would prevent 'data rate creep' that could prevent success in the system design. Thus, 164 kbps is viewed to be our final required throughput, although this includes 16 kpbs of control, which is effectively overhead.

1.2.6 UHF Operation

At the physical layer, the UHF band is proposed for the RF signals. UHF is selected primarily due to the bandwidth and propagation trade-off advantages that it offers. UHF band communications suffer from significantly less congestion and interference than in the case of VHF band communications.

UHF is a mature technology already in wide use for communication systems. This band provides the potential for wider bandwidth and inherently implies frequency reuse due to higher signal attenuation [1]. Higher signal attenuation is, however, clearly a disadvantage in other respects; UHF is not intended for long-range applications.

Finally, and perhaps most importantly, a significant portion of the 900 MHz frequency band is already designated for land-mobile use. As a result, existing knowledge can be leveraged in fielding such a system and regulatory concerns would be minimized. For consistency, 900 MHz will be used as the center frequency for further discussions.

1.3 Requirement Summary

In summary, given these criteria, the system design must provide for the following features and characteristics:

- RF based system operating in the UHF band (at 900 MHz) utilizing a common transceiver platform;
- 25 mobile users supported over a 20 km by 20 km area;
- rapidly deployable, dynamic, and survivable multi-hop architecture with no fixed infrastructure; and,
- 164 kbps data rate for multimedia service with appropriate QoS bandwidth and delay guarantees for integrated point to point voice, video and data communications voice, specifically:
    - four half duplex voice channels of 8 kbps, leading to a 32 kbps requirement;
    - three half duplex video channels of 32 kbps, leading to a 96 kbps requirement;
    - two low speed data channels of 2 kbps each;
    - 2 medium speed data channels of 8 kbps each; and,
    - a control channel of 16 kbps (overhead).

The question of broadcast capabilities will also be addressed, but will this is not specified as a system requirement

1.4 Thesis Organization and Contributions

This thesis will address the majority of high level issues that a network designer must consider for mobile multimedia networks in a distributed environment. These issues are discussed in the context of the operational requirement defined in Chapter 1.

A channel model based upon the Hata path loss formula will be developed in Chapter 2. This will employ log-normal shadowing to provide a more accurate simulation of real-world conditions. This non-linear channel model provides a novel approach to channel simulation.

A discussion of connectivity methods for distributed networks will follow in Chapter 3, with an emphasis on linked cluster architectures that have figured prominently in recent research. It will be shown that these techniques are not necessarily required to achieve acceptable network operation.

Chapter 4 focuses on the major multiple access techniques available for this design. Discussion concentrates on TDMA and CDMA solutions leading to an examination of a version of CDMA termed slotted CDMA. Based upon the requirements stated above, a novel use of such slotted CDMA is proposed with a view to providing the desired services with required QoS in a distributed environment.

In Chapter 5, three variants of slotted CDMA are examined for use in this network and the most appropriate is selected. With the multiple access technology defined, specific issues related to network operation and control are investigated. This includes further modification of the system's actual data transport requirements to a fixed data structure and definition of the network's integrated control channel. This in turn allows for discussion of critical issues including path determination, cell routing, scheduling and resource or link reservation.

Chapter 6 explores the integration of the aforementioned network elements and the challenges related to bringing the network's parts together. In addition, desirable enhancements to the network's capabilities through the use of specialized features such as data broadcast and a voice sub-network are discussed.

Chapter 7 provides an overall summary of the network design detailed in the preceding chapters. Chapter 7 concludes with recommendations for future research and refinement in the area of wireless, survivable, integrated multimedia networks.

2.0 Channel Model

2.1 Introduction to Channel Modeling

Channel modeling is an absolutely critical area that must be addressed when attempting to formulate a realistic network design. However, channel modeling is often simply ignored or at best replaced by a basic radio-range based approach [3],[11], [12], [13]. In such a model the transmitted signal is attenuated in accordance with a square law distance loss wherein the ability to receive is judged on a simple binary, go/no go basis. However, this is somewhat simplistic and unrealistic, and precludes the simulation of scenarios such as the uneven connectivity described below in Figure 2.1 to occur. Indeed, Gerla has suggested that more realistic modeling should be included in future efforts [13].

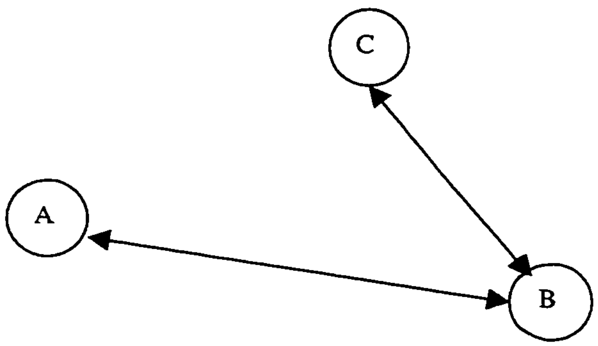


Figure 2.1: Uneven connectivity

In Figure 2.1, node A is physically considerably closer to node C than node B. If a purely range based approach was utilized, we could assume that because A is connected to B it should also be connected to C. However, in reality, conditions exist whereby it would be possible for node A to be capable of contacting node B but not node C, thus resulting in uneven connectivity. This would be the case if, for example, a large physical obstruction separated nodes A and C but not A and B.

While Figure 2.1 cannot be modeled in a simple radio-range scenario, it can be simulated through the addition of other channel model components such as log-normal fading. It is true that, in general, the attenuation from node A to C will be less than from node A to B. However, when a Gaussian random variable is introduced to adjust both of these calculated figures, it then becomes possible to have the situation outlined above to occur.

2.1.1 Channel Modeling Scale

RF channel modeling is fundamentally approached in three different categories. First, at the macroscopic level, path loss is related to the range between the transmitter and the receiver. Second, large scale fading occurs due to motion over large areas where terrain features vary. Finally, small-scale fading is observed due to small changes in position and multipath propagation.

At the macroscopic level, a range-based approach can be applied. Typically such a regime is calculated on a fourth law basis, although there may be some variation in the exact exponent used. Depending upon the environment the path loss can be inversely proportional to the distance of separation raised to a power between 2 and 5 [4]. If the channel simulation were to be entirely based upon such a model, it would be possible to simply draw range circles around system terminals to determine system connectivity, and in many approaches this is exactly what has been done. Unfortunately this does not accurately represent the connectivity observed in actual networks. Therefore other channel model components should be considered.

Large scale fading is influenced by terrain features and can cause phenomena described as "log-normal fading", named for its relationship to a Gaussian statistical approximation. Log-normal fading can be applied to express variation in average power set about the median level determined at the macroscopic stage. This allows for real-world variations to be described as illustrated in Figure 2.1.

Signal variations in the local micro-environment can be described as following a Rayleigh probability density function, and is therefore known as Rayleigh fading. Rayleigh fading typically results from multipath reception where no line of site signal component exists. While not insignificant, as demonstrated by Viterbi it is not a critical requirement for this channel model [21]. To keep the model focussed and simple Rayleigh fading is therefore omitted from further discussion. It should be noted that CDMA communication systems (to be explored in Chapter 4) are largely unaffected by Rayleigh fading.

In any event, to keep the channel model realistic some form of fading must be anticipated and accordingly log-normal shadowing will be pursued to allow uneven connectivity to occur. The data produced by the channel model suggested herein will demonstrate that such a simulation can be produced in a relatively straightforward fashion without adding significant complexity.

2.2 Free Space Path Loss

A free space path loss relationship provides the simplest channel model, essentially following a simple range related fourth power law relationship. In other words, as the path length between two points doubles, the path loss between the same two points is two squared or, for each 3 dB increase in path length, path loss increases by 6 dB. This path loss model is based upon the following power relationship and holds true if R is large with respect to the antenna heights:

$$P_r \cong P_t\, G_t\, G_r\, (h_b h_m / R^2)^2$$

where h_b and h_m are the base and mobile antenna heights, G_t and G_r are the antenna transmit and receive gains, and R is the range [8].

The accuracy of the relationship is however called into question when exploring the topics of angle of incidence and conductivity of the terrain over which the information is being transmitted. Yet more real-world factors can be taken into account through the application of diffraction models which allow for obstructions such as buildings. Diffraction models will not be pursued here in the interests of simplicity, however a more comprehensive approach can be applied without adding great complexity through the application of the Hata-Okumura model.

2.3 The Hata/Okumura Approach

While a free space path loss approach provides a good starting point for a channel model, such a technique does not take into account geographical features or atmospheric perturbations that can have a major impact on signal propagation.

Fortunately, empirical methods for more realistic simulation and prediction of path loss have been developed for wireless applications. Two of the most commonly used models are the Hata-Okumura model and the Walfisch-Ikegami model [4]. The latter is a combination empirical-deterministic model, primarily intended for cellular communications in an urban environment. Of more relevance in this case is the Hata-Okumura approach which is also widely used and offers relatively good accuracy with little added complexity [5].

The Hata-Okumura model is derived from Okumura's technical report wherein a reference level of median attenuation is calculated for certain fixed parameters [6]. These parameters, including the base antenna height, mobile antenna height, frequency and terrain classification, are then adjusted to reflect the specifics of the scenario.

In 1980, Hata modified Okumura's work to create simple mathematical formulas [7]. These new formulas make it possible to quickly calculate path loss for a variety of environments from heavily built-up areas to wide-open environments. The resulting Hata-Okumura modeling methods are heavily quoted and referred to in the literature.

2.3.1 Hata Path Loss Equation

Hata proposed the following path loss relationship:

$$L_p\,(dB) = 69.55 + 26.16 \log f_c - 13.82 \log h_b - a(h_m) + (44.69 - 6.55 \log h_b) \log R$$

Where

f_c is the carrier frequency between 150 - 1500 MHz
R is the range between 1-20 km
h_b is the base station antenna height (30-200m)
$a(h_m)$ is the mobile station antenna height correction factor

In this case simplifications are required. First, due to the distributed nature of the system there are no base stations. The mobile stations are all identical and no antenna height approaches 30 m. However, as all stations are identical, the relative impact of this criteria is consistent and therefore the overall effect on the approximation is negligible.

In the case of stations within 1 km of each other, it is very safe to assume that connectivity will be achieved. In such cases there is no real reason to quantify path loss as it is only utilized in later stages to ascertain whether or not connectivity can in fact occur. This criteria is therefore also discounted.

Several modifications are also described by Hata to account for different terrain categories, specifically open, suburban and different urban classifications (medium-small and large city). Further evidence of the utility of this approach can be found in European Cooperation in Field of Scientific and Technical Research (EURO-COST) application reports 207 and 231. The COST 231 report empirically applies the Okumura/Hata approach, tailored to PCS micro-cellular systems in the 1500-2000 MHz range.

The Hata terrain adjustments are based upon the spread between the 10% highest elevation and the 90% highest elevation along the path. However, while many terrain profiles have the same 10%/90% spread, it is unlikely that they will have the same shadowing losses. For that reason, these adjustments can only be viewed as an approximation. For greater accuracy, shadowing adjustments could be generated by the application of diffraction theory to the terrain data. Clearly, this adds a great deal of complexity.

For this work, actual terrain data is not considered for use in the model and the model is not derived from an actual 20 km by 20 km terrain grid. The model is in fact the opposite: an arbitrary channel simulation. Therefore, it is possible to assume that the Hata approximation is an exact mathematical interpretation of the propagation over a hypothetical piece of terrain. It therefore meets the requirement of the channel model for that piece of terrain.

2.4 Log-Normal Shadowing

The second regime of channel simulation consists of log-normal shadowing which, via Gaussian statistical approximations, provides a realistic portrayal of actual propagation fluctuations observed in nature. Thus as distance increases, we expect to observe a general decrease in signal strength with spikes and lulls about the mean curve.

Figure 2.2 plots signal strength versus radial distance from a given transmitter [8]. It illustrates the real world fluctuations that are observed about the mean theoretical propagation curve. This can be due to numerous factors, including geographical features or peculiarities, and atmospheric conditions and perturbations.

Figure 2.3 displays a mathematical method to simulate these fluctuations via log-normal shadowing about a signal strength curve as described above [9]. These calculations will now be added to the channel model

This log-normal shadowing can be introduced to the channel simulation model by determining the standard deviation (σ) which defines the Gaussian statistical variation for the model and applying this variable to the general path loss formula defined above. This is done for each path loss calculation.

The most common standard deviation referred to in the literature for log-normal fading is $\sigma = 8$ dB [8, 21]. Accordingly, this figure is selected to produce the Gaussian adjustment to the Hata path loss curve for $f_c = 900$ MHz.

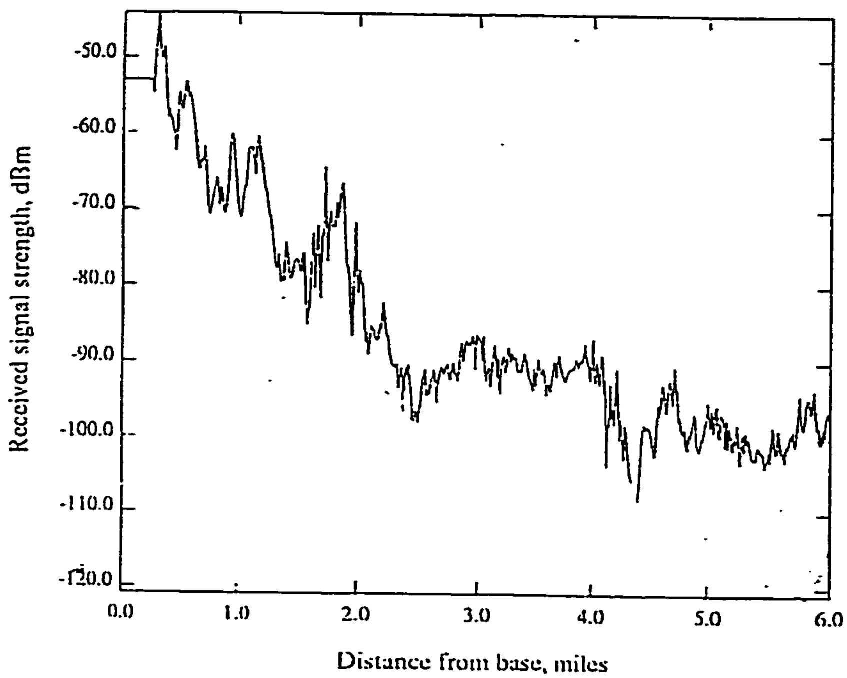


Figure 2.2: Variation to Median Attenuation [8]

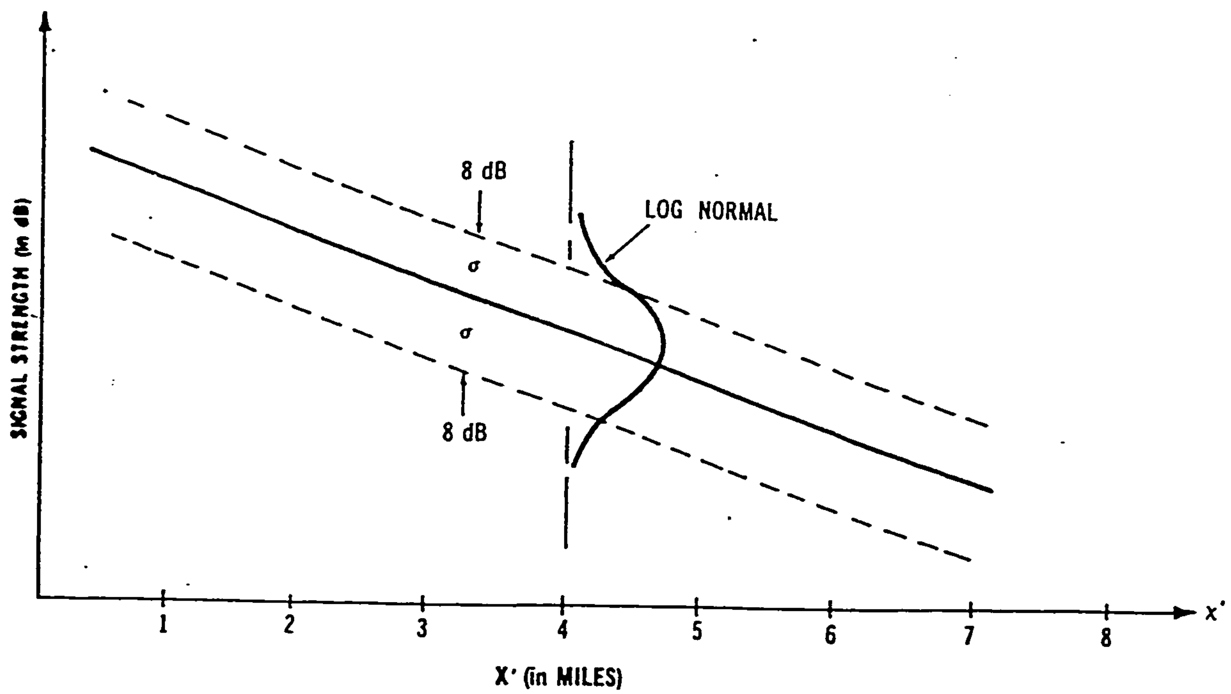


Figure 2.3: Log-Normal Shadowing [9]

2.5 Hybrid Channel Loss Model & Simulation

Using the methodology outlined in Section 2.4, a hybrid channel loss model can be produced for use in network connectivity simulations. This will produce baseline data for the connectivity maps upon which network observations can be based.

For this research the channel loss model has been developed through the use of a Matlab script which calculates the simulated path loss between each pair of points in the 25 user network. It does so by randomly distributing 25 x-y points over a 20 km by 20 km grid through the use of a uniform normal regime. The distances between the points are easily calculated.

Once all inter-point distances have been determined, the Hata empirical path loss formula is used to obtain the mean path loss curve for the parameters determined for this network and the distances just calculated. Next, this curve is adjusted by the addition of a Gaussian random variable with 8 dB standard deviation, resulting in an adjusted Hata path loss figure representing the path loss for each pair of points in the network.

With such data it becomes possible to plot the relative locations of the network terminals. If a cut-off threshold for path loss is applied, it becomes possible to determine network connectivity and produce a graph of this as well. It is absolutely critical to note that these connectivity maps are determined through the use of the channel model in tandem with this cut-off threshold.

The cut-off threshold figure is initially and arbitrarily set at 140 dB for descriptive purposes, and determines the path loss that can be incurred while still maintaining an acceptable SNR for a given data type and its inherent required bit error rate. The actual number used will depend upon the transmit power, the interference power of other simultaneous transmissions (in the case of CDMA) and the noise level which is based in part upon the transmit bandwidth.

However, given the various classes of service that must be supported and the initial data requirements identified in Section 1.2, it will be necessary to provide mathematical reconciliation between the two, and not merely assume that they are mutually supportive in the preliminary format. This exercise will be carried out in Chapters 4 and 5.

The Matlab script code developed and employed for this work is listed in Appendix 1 for the 140 dB threshold example. Figure 2.4 is an example connectivity plot produced by the Matlab script which illustrates the network's configuration map for one given set of random data. This data produced is attached in Appendix 2 and is used as one of the sets of baseline data for further discussion.

The data produced by the Matlab script includes:

- the xy coordinates of the 25 points;
- a matrix of the distances between all points;
- a matrix of the Hata path loss between all points (in dB);
- a matrix of the adjusted Hata path loss between all points (in dB);
- a binary array of clusterheads;
- a binary matrix indicating which nodes are subordinate to which clusterheads;
- a binary matrix detailing which nodes are connected to each other;
- a matrix indicating which nodes are one hop gateways between clusterheads;
- the total number of links between nodes; and,
- the average number of links per node for the network.

As mentioned, several baseline data sets are used for discussion in this thesis. For example, the terminal distribution listed in Appendix 2 is henceforth referred to as the 'α' data set (in this particular example, as applied to a 140 dB threshold). Each data set is based upon a specific seed key used by the Matlab *rand* and *randn* random number generator functions used to determine point distribution. By using a small set of seed keys it is possible to provide repeatable results and common network distributions for different threshold values. In addition to the α data set, several other baseline sets will also be used, namely the β, γ, δ, and ε data sets.

With the channel model established based upon the Hata-Okumura approach, the focus of this effort shifts to the formulation of a network architecture capable of supporting fully distributed and survivable operation.

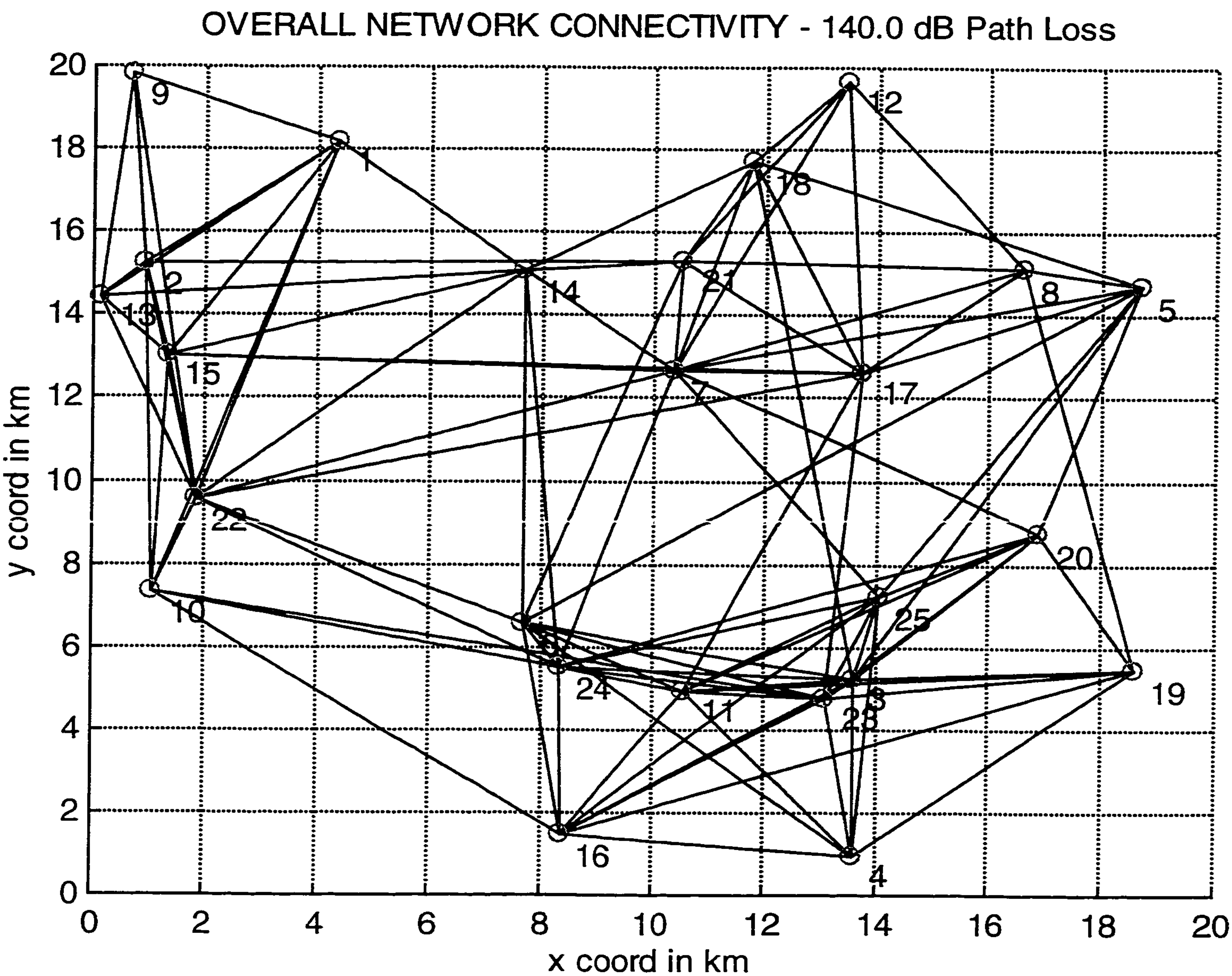


Figure 2.4: Sample network connectivity map - baseline α data set,
140 dB sustainable path loss

3.0 A Distributed Network Architecture - the Linked Cluster

3.1 Introduction: The Implications of Survivability

One of the key arbitrary design considerations made at the beginning of this work was the requirement for survivability. This requirement may appear innocuous, however in terms of the network design it immediately rules out one of the central tenets of cellular network design: the base station.

In order for a network to be survivable, it must be possible to remove any given node without significantly eroding the network's coverage or performance. In a cellular network, the removal of a base station would result at a minimum in a 'dead zone', an area in which mobile subscribers would be out of communications. Therefore, base stations cannot be employed.

In a truly survivable system, the subscriber's equipment is the network. The removal of one node simply results in the removal of that node from the communications network. In other words, the effect to the network would be much more localized than the loss of a base station in a cellular network.

Another key requirement for distributed mobile networks is the ability of the topology to adapt to change and movement. The topology must be dynamic to reflect the changing distribution of nodes on the ground. Furthermore, knowledge of the topology must be disseminated throughout the network. Updates must be provided periodically to allow for the relative movement of nodes in the network, thereby permitting the allotment of new routes for delay sensitive circuits affected by the topology changes. This will ensure that connectivity changes are taken into account when nodes set-up and operate communication links.

Having made these decisions, many other related considerations remain to be clarified. For example, how distributed or ad-hoc will the network really be? Will the self-organizing network appoint some stations to perform hierarchical tasks, making them more significant to the network, or will the network truly be a conglomeration of equals? How often should the network be updated? What type of control will be used?

The latter two topics will be discussed in following chapters while an examination of the ad-hoc nature of the network and a recommendation provided for the overall network topology is provided below.

3.2 The Linked Cluster Architecture

Numerous sources propose and describe a concept known as the Linked Cluster Architecture (LCA) [3,10,11,12]. In these papers the LCA is used to form the distributed packet radio network topology. Originally applied to a US Navy low-rate application known as the HF Intra-Task Force (ITF) Communication Network [12], the LCA was proposed for a variety of experimental distributed wireless systems.

Application of the LCA produces a topology wherein all network nodes have been organized into clusters, with a clusterhead identified for each and every node. In this framework, each node is assigned to only one cluster although it may have communications established with several clusters.

The architecture is itself produced through the application of an algorithm such as the lowest-ID cluster algorithm, the highest ID cluster algorithm or the highest connectivity algorithm. Network stability was identified as one of the most important characteristics to be evaluated when selecting a connectivity algorithm. The lowest ID cluster algorithm was selected in [3,10,11,12] as it was identified to provide the most stable topology (i.e. the topology with the fewest changes when node motion is added). It is therefore selected for further investigation here.

3.2.1 Lowest ID Cluster Algorithm

The lowest ID cluster algorithm is illustrated in Figure 3.1 on the following page. The algorithm effectively steps through each node in a sequential order beginning with the lowest ID node. Connectivity is determined for each node and, if a cluster has not been identified to the terminal in question, its number is assigned to the cluster. All other nodes connected to that node are then also assigned the same cluster ID number. If the node under examination has already been assigned to a cluster number, the node number is incremented and the loop restarts. This is repeated for all distributed stations.

If the lowest ID cluster algorithm is applied to the sample 140 dB sustainable path loss, α data set (with the connectivity mapped as in Figure 2.4), the hierarchical scheme illustrated in Figure 3.2 results. In Figure 3.2 Clusterheads are identified with an asterisk superimposed over the terminal symbols (e.g. nodes 1, 3, and 7). It is observed that nodes do not necessarily cluster with those in the immediate vicinity. Indeed, node 18 (located in the upper right hand side) is closer to node 7 but is assigned node 3 as a clusterhead.

The LCA implementation results in different node classes. Clusterhead stations assume various higher network management responsibilities and are specifically responsible for resource allocation within their clusters. Gateway nodes are used to move traffic between various clusterheads. All other terminals are termed ordinary nodes.

However, before selecting the LCA for application in this case, it is more fundamentally sound to examine the perceived benefits of the LCA and determine if it is really required.

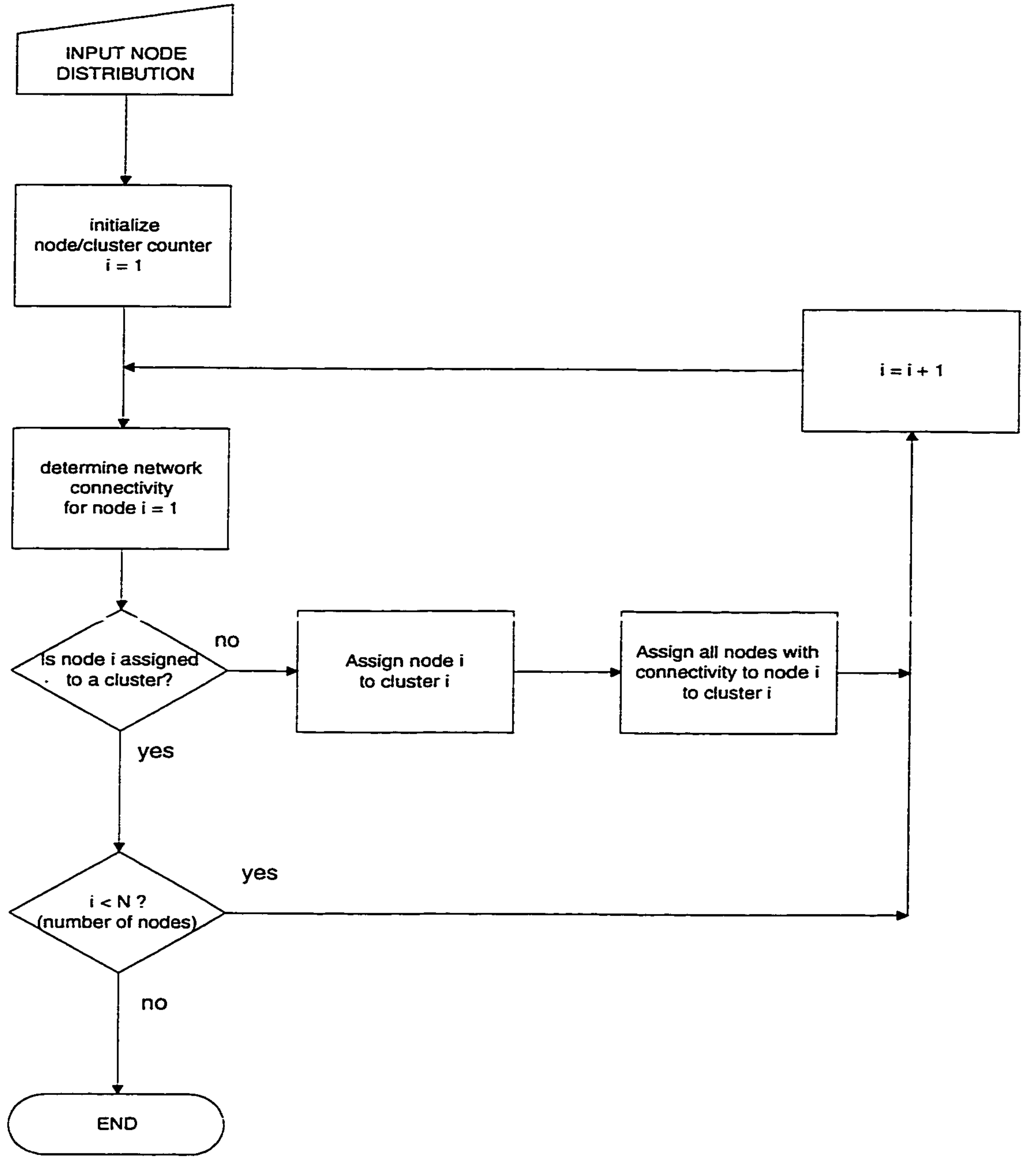


Figure 3.1 Lowest ID clustering algorithm

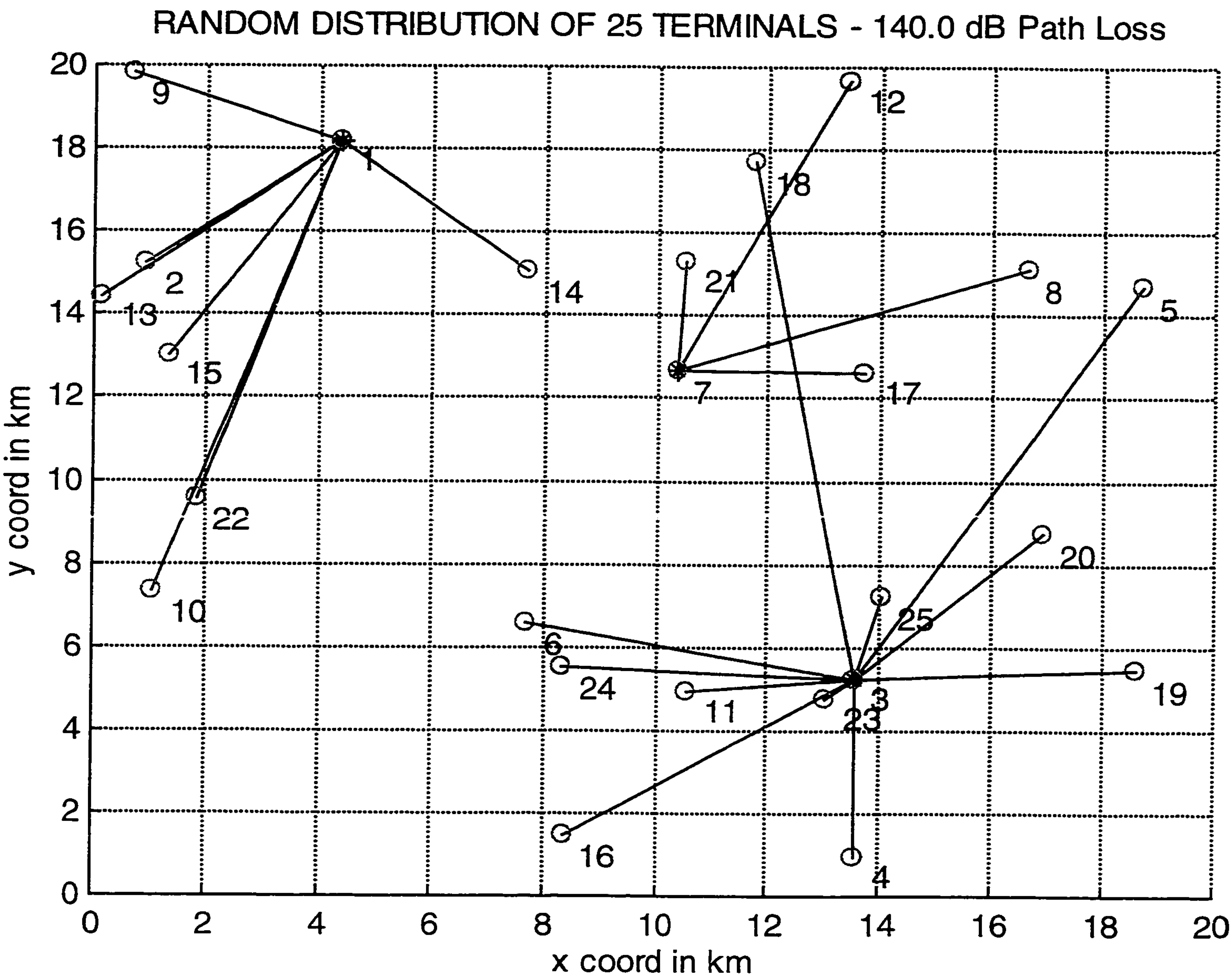


Figure 3.2 Clustering example – lowest ID algorithm
140 dB sustainable path loss, α data set

3.2.2 Clusterheads

In the linked cluster architecture, clusterheads are selected through the clustering algorithm and are then assigned to perform network functions. Essentially, the clusterheads must serve as 'local coordinators'. They act to resolve channel scheduling, perform power measurement and control, maintain time division frame synchronization and enhance the spatial reuse of time slots and codes. Data flow between different clusters flows through intermediate nodes and gateways between two clusterheads.

The addition of clusterheads does unfortunately introduce significant bottlenecks which can present considerable difficulties, and in a worse-case situation could shut the network down [13]. When considering that some QoS requirements will require the establishment of Virtual Circuits, it is evident that the network should assign as few nodes as possible to the route. If clusterheads must be used, then additional nodes are dedicated to a task and efficiency will suffer as a result. Given the limitation of bandwidth available for our system, longer routes must be avoided.

In Figure 3.2 a total of three clusterheads are in place. Most VC traffic will take place between nodes that are not contained in the same cluster. This is simply due to the fact that most nodes are not in the same cluster. This can result in very inefficient routes between terminals.

For example, if a video circuit were to be established between nodes 5 and 8 (located close to each other in the upper right hand corner), the LCA would require that the transmission path take place between nodes 5, 3, 7 and 8. With the majority of communications taking place between nodes in different clusters, the creation of bottlenecks is inevitable.

Just as clusterheads can lead to choke points, so can gateways. Clusterheads 1 and 7 are connected by three one-hop gateways. These gateways would have to carry all intercluster traffic between the two nodes, in addition to their own traffic. This is problematic when bandwidth availability is very limited. Of even greater concern is the lack of one-hop gateways between clusterheads 1 and 3. Any intercluster communications between these two clusters would result in the addition of considerable overhead.

Furthermore, some of the other concerns that clusterheads are designed to address may not be critical in this network design. Synchronization will be a very important element in this design, but can be provided via GPS. Spatial reuse and code assignment will also be shown to be minor concerns in this case.

More recent work identifies the bottleneck that clusterheads introduce and proposes the adoption of a fully distributed approach for cluster formation and intra-cluster communications [13].

3.3 Adaptive Clustering

Under the Wireless Adaptive Mobile Information System (WAMIS) initiative, several mobile, multimedia, multihop (M^3) architectures have been developed [13]. Lin and Gerla's recent reexamination of the cluster approach for WAMIS has discounted the requirement for clusterheads. Due to the problems related to information flow through clusterheads discussed in the previous section, they propose instead the concept of adaptive clustering.

As previously described, clusterheads introduce information bottlenecks of questionable value, particularly if the transmitting nodes and the receiving nodes are close together but assigned to different clusters. With the adaptive cluster approach, cluster formation remains as previously proposed. Now however, all system nodes truly are equal in terms of the duties they perform. The objective of the adaptive clustering algorithm is to "find an interconnected set of clusters covering the entire node population" [13]. More specifically the system topology is divided into clusters with independent control.

The adaptive cluster approach essentially provides the required QoS to nodes in two steps:

1. the network is partitioned into distinct clusters to allow for controlled, accountable bandwidth sharing in each cluster (assumes frequency division or reuse); and,

2. the establishment of virtual circuits (where required) with a related QoS guarantee [13].

Clearly, cluster size is dictated by propagation conditions and the power level transmitted by terminals. In [13], transmission power is fixed and assumed to be uniform across the network. Within each cluster, as before, nodes can communicate with each other in at most two hops.

Lin and Gerla state that it is possible to adopt a fully distributed approach for cluster formation [13]. However, the cluster formation methodology remains unchanged from the previous proposal from Gerla [3]. As before, the clusters can be constructed based upon node ID and the concept of the lowest ID algorithm can be readily applied [3,10,11,12]. The term 'repeater' is applied to nodes that relay packets from one cluster to another. In Figure 3.2 above, node 2 for example is a repeater.

Lin and Gerla go on to describe in great detail the method by which QoS guarantees are established and maintained through TDMA. Although the actual data transmission mechanism is not discussed in any meaningful detail, it can be interpolated that the network can inherently build routing tables based upon each node's knowledge of its one and two-hop neighbors.

It is also possible to for terminals to concurrently build individual routing tables based upon the dissemination of connectivity information throughout the network, without the implementation of a linked cluster architecture. Leiner describes a simple mechanism that can be used to permit each terminal to construct routing tables [1]. This concept will be discussed at length in Chapter 5.

Clusterheads are not critical to this design and therefore are not employed for routine use in this network. However, it is not essential to totally disregard the utility of clusterheads or topologies based upon the lowest ID algorithm. As previously identified, the ability to broadcast information to all network terminals is desirable. Transmission of broadcast traffic to all stations in the network may in fact be most easily accomplished via clusterheads. Therefore, this concept will be further explored in Chapter 6.

4.0 Channel Access

4.1 Multiple Access and Packet Radio

The most central theme in the concept of the network is that of resource sharing. In the case of a packet radio network, this translates into the ability to share bandwidth in an efficient manner, dependent upon some multiple access technique.

Efficiency is critical in view of the very limited bandwidth that is available for wireless applications. This is more difficult to address in the wireless world, even in cases where base stations permit some form of centralized control. The added requirement to operate under conditions of fully distributed control wreaks havoc with the ability to efficiently and dynamically divide available resources to meet the desired traffic load. With no central omniscient control station it is difficult to codify a methodology that will optimize the allocation of network resources. Distributed control imposes significant limitations and it is therefore necessary to sacrifice some measure of efficiency in favour of the requirement to be survivable.

The requirement of a multimedia network calls for the provision of various traffic types, consisting of delay sensitive voice and video circuits in addition to various speeds of delay insensitive data. To provide these services implies the requirement to provide a guaranteed quality of service (QoS). In view of the limitations related to distributed control, the network's control structure alone cannot provide these guarantees. The network's data structure must assist in providing suitable QoS.

Related to QoS provision and network control is the issue of bandwidth sharing and scheduling. Will transmission schedules be determined in advance? Will contention algorithms be used to allow for a quasi 'bandwidth-on demand' approach in which the channel is sensed for activity and transmissions are made based upon bandwidth or receiver availability? These issues will be discussed in greater detail in Chapters 5 and 6 after the multiple access technology has been selected.

The sharing of bandwidth in RF applications has traditionally been approached in three distinct categories: Time Division Multiple Access, Frequency Division Multiple Access and Code Division Multiple Access. In reality, more options are available if one considers other approaches that combine some or all of the three basic methods.

TDMA and FDMA in their pure form are narrow band topologies. In TDMA a common bandwidth is sliced in time to permit multiple users, either on a fixed or demand access assignment basis. In FDMA, users are assigned individual frequency ranges, which requires more complex receivers. Transmission overlap in both cases will usually result in the destruction of all data. CDMA on the other hand is a spread spectrum approach, which in contrast, allows simultaneous message transmissions over the same bandwidth.

From a theoretical point of view, all three methods provide equivalent radio capacity [21]. In reality however, system requirements and other practical considerations for network implementation will favor one technique over another. The reemergence of CDMA as a solution for mobile systems has created a debate about which technique is preferable: TDMA or CDMA.

Opinions and results on this topic vary significantly. Papers exist that show CDMA to be less spectrally efficient than FDMA or TDMA while other results demonstrate that the efficiency of spread spectrum systems can exceed the narrow band schemes by a factor of five [21]. Due to the wide range and simplicity of simulation models used to reach these disparate conclusions, no substantive answer has emerged. In this thesis the requirements listed in Chapter 1 will assist to determine the preferred multiple access technique.

FDMA will not be discussed as a separate option. Although FDMA is the enabling technology for AMPS in North America and TACS in the UK, this method is not widely quoted in the literature identifying future wireless approaches. Emerging mobile wireless systems consist primarily of TDMA and CDMA systems. Therefore these two techniques, and a version of CDMA termed slotted CDMA, will be explored for use here.

4.2 TDMA

In its most basic form, Time Division Multiple Access provides for discrete, scheduled connectivity throughout a given network. TDMA is presently in very widespread use for mobile systems, particularly cellular and is also widely quoted in the literature primarily due to its simplicity. However, this simplicity is not easily adapted to the more complex mobile multimedia environment.

TDMA allows for basic technological solutions and simple receivers. TDMA signals are robustly disjointed [20] if properly synchronized and the demands on power control are low. It should be noted that synchronization requirements for TDMA are not truly viewed as a disadvantage because virtually all proposed architectures will have to utilize some form of network synchronization.

However, for all its simplicity, TDMA technology inherently introduces one major disadvantage: flexibility. TDMA does not easily reconfigure schedules to allow for variable rate traffic. Given the extremely wide variance in bandwidth requirements for the nodes in our network (ranging from no requirement, to time sensitive voice and video, to non-sensitive data) this can hamper efficiency, particularly when handling bursty traffic and unbalanced load requirements [1]. TDMA is most suitable for predictable, constant services such as that required by a dedicated control channel.

Some measure of flexibility can be applied to a TDMA network via the use of dynamic slot allocation. Unfortunately, this then requires the application of scheduling algorithms, which tend to become complex in a distributed environment [1].

TDMA requirements and restrictions vis-à-vis collisions are much more strict than the case for CDMA. If two packets collide in TDMA, they are altogether lost. This is referred to as 'zero capture' [1]. Therefore, for TDMA architectures two nodes need to be aware of the possibility of interfering with each other, and cannot transmit at the same time if they are in the same radio range. Thus, further constraints must be placed upon the network infrastructure to prevent such collisions from occurring. This requirement is much less restrictive in CDMA systems where it is possible for terminals to transmit simultaneously without destructively interfering with each other, unless the sum of the spread interferers and the background noise exceed a certain threshold.

Another key disadvantage with TDMA systems in the mobile environment is the high peak to average power ratio required for the terminals. Power requirements are greater than for CDMA systems, and this would typically result in either larger batteries or shorter battery life, neither of which is desirable in a mobile system, particularly a handheld unit.

4.2.1 System Requirements Applied to TDMA

The effective data rate that must be supported by the design, regardless of the multiple access technology, is 148 kbps plus 16 kbps for control. As previously stated, in

TDMA only one terminal can transmit at any one given time. If multiple hops are prevalent for virtual circuits that are delay sensitive, then the overall actual data rate would be higher, perhaps much higher than 148 kbps. This is because, as shown in Chapter 1, multiple hop VCs require multiples of the raw VC data rate; a three hop 32 kbps video circuit was shown to really require 96 kbps.

Before connectivity can be estimated, it is necessary to determine what the acceptable path loss or cut-off threshold would be for a TDMA network. Several options will be explored to provide a range of answers, which will then be used in conjunction with the channel model to obtain connectivity diagrams. These diagrams will permit comparison of the various options.

Two basic approaches can be taken with respect to scheduling services through the network. The most simple, but least efficient solution is to allocate a fixed portion of the bandwidth to each of the 25 stations in the network. This is inefficient because the terminal is allocated the bandwidth, regardless of whether or not it is needed. On the other hand, this action minimizes the requirement for dynamic control and scheduling, both of which are problematic in a distributed system.

Another approach is to provide for a pool of data slots that can be obtained and reserved by network nodes for the transmission of traffic. If one ignores for the moment the problems related to the complex control and allocation of resources required here, it is possible to conclude that this will be much more efficient than in the fixed allocation case.

In either situation, it is possible to examine a range of capabilities. On one extreme, a low throughput system can be designed that will allow for greater network connectivity but poor availability. One can also over-design the system to ensure that all required traffic can be carried at any time through any node.

For simplicity, three situations will be explored in this case. Specifically:

- a fixed assignment scheme providing 32 kbps of bandwidth to each node;
- a demand assignment type system with a total of 328 kbps segmented in various frame sizes defined below; and,
- a demand assignment system with 526 kbps segmented in various frames sizes.

The extreme situations on either end of the spectrum are not feasible and are therefore discounted. For example, a demand assignment system with a throughput of 148 kbps would only be of use if the vast majority of nodes in the network were connected by one hop links. This is not likely as evidenced by the connectivity diagrams which follow. The other extreme example is a system enabling all traffic to flow through one node. As this would exceed 3 Mbps it is not a valid application for mobile TDMA technology.

The 32 kbps/node fixed system requires an overall data rate of 816 kbps. The system would allow for each node to route either one video link, or a combination of voice and data links. Thus, the system will likely be hard pressed to provide for three simultaneous video channels. The connectivity map provided by the channel model will illustrate this.

The 328 kbps demand assignment system is based on the ability of the network to provide the following:

- six 32 kbps slots (which would support three video links if they were for example 3, 2 and one hop in length);
- eight 8 kbps slots for voice (e.g.for four VCs: 3, 2, 2 and 1 hop in length);
- four 8 kbps data slots (two paths of length 2);
- four 2 kbps data slots (two path lengths of two); and,
- 32 kbps for control.

The 526 kbps demand assignment system provides the ability to support longer paths to support the various circuits through the network:

- ten 32 kbps slots (e.g. for three VCs: 4, 3, and 3 hops in length);
- thirteen 8 kbps slots (e.g. for four VCs: 4, 3, 3 and 3 hops in length);
- seven 8 kbps data slots (e.g. 4 and 3);
- seven 2 kbps data slots (e.g. 4 and 3); and,
- 32 kbps for control.

The two demand assignment options include a 32 kbps control channel. This is a very conservative estimate of the added control requirements related to the implementation of the demand assignment algorithm in a distributed environment.

4.2.2 TDMA Link Budgets

Link budgets for these three options have been calculated, and the results are shown in Figure 4.1. The link budgets compare the ratio of the transmitted power minus the sustainable path loss (or cut-off threshold), to the bit rate multiplied by the system noise temperature and Boltzman's constant, such that a margin of 10 dB is maintained. The system noise temperature is assumed to be 2000°K. The calculations are based on four different levels of transmit power ranging from 0.1 W to 2 W. Two Watts is assumed to be the maximum feasible for a mobile system operating away from a power source (the terminals are assumed to be man portable as opposed to vehicle mounted).

TDMA link budget:

$$\frac{P_t - \text{cut-off threshold}}{1.38 \times 10^{-23}\ \text{W}\cdot\text{s/K}\ (2000\text{K})\ (R_b)} \geq 10\ \text{dB}$$

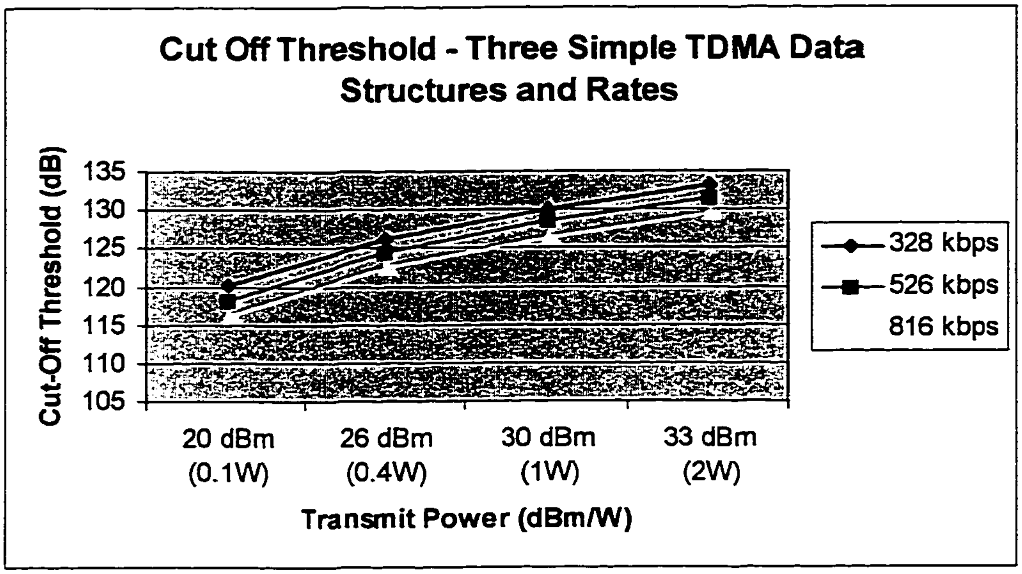


Figure 4.1: Cut off threshold – three simple TDMA data structures and rates

In terms of capacity, the least capable of the three systems is the 328 kbps network. With the lowest bit rate, it also predictably allows for a higher cut-off threshold. Therefore, of the options investigated, this configuration with the highest transmit power (2 W) provides the most comprehensive network connectivity because it yields the highest, or most permissive, cut-off threshold. If we utilize the α data set the resulting connectivity map is provided in Figure 4.2a. Figure 4.2b provides the connectivity map for the 816 kbps case with a 2 W transmitter for comparison.

Clearly, the connectivity for the 328 kbps network is better than for the 816 kbps network. However, it is evident that neither configuration provides sufficient connectivity to allow for even basic network operations. A network with 25 nodes could have a maximum of 300 links if all nodes were connected with each other. The 816 kbps network provides just 34 links and the 328 kbps network provides only 58 links of the possible 300 connections as seen in Figure 4.2a.

In Figure 4.2a the network does provide connectivity to all nodes. If no video connections were required, then the data capacity might be sufficient to support basic operation. However, the requirement to route three different video links will overwhelm the network because only six slots are available. This would support at most three two-hop connections or a three hop, two hop and one hop configuration for the three video VCs.

Figure 4.2a connectivity for α data set, 328 kbps network, 2 W tx power

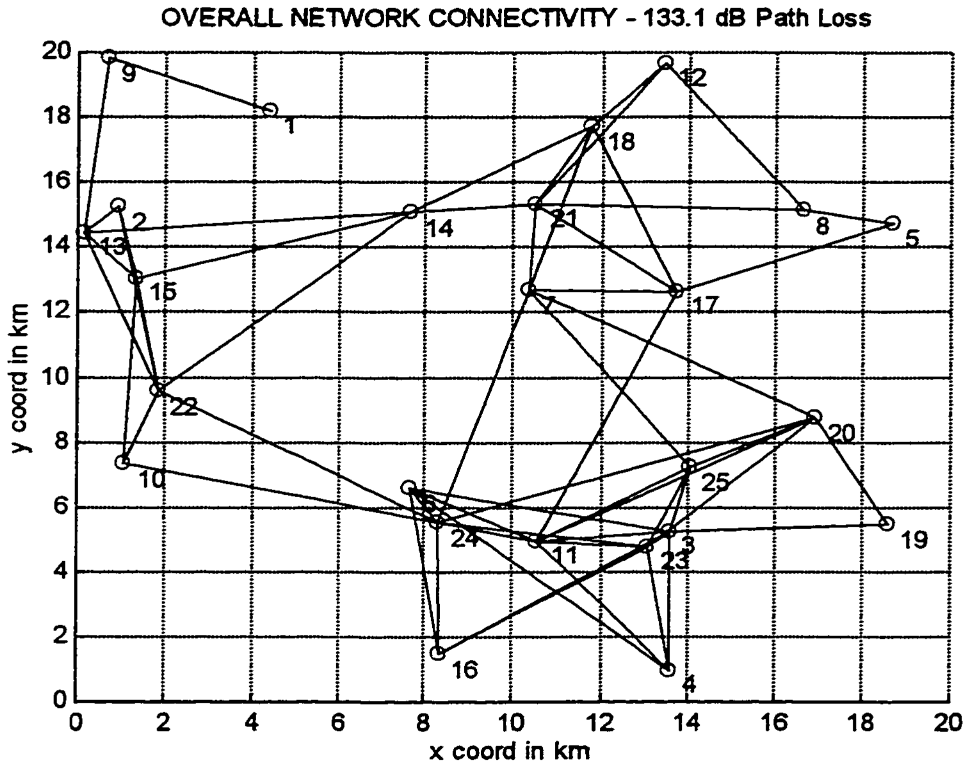
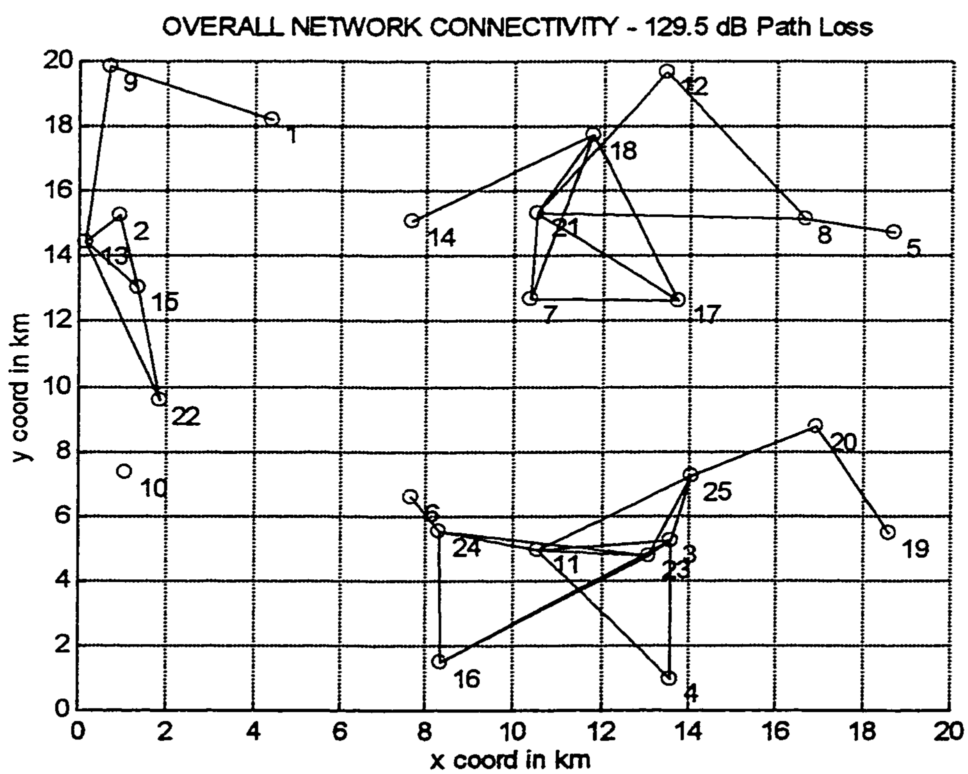


Figure 4.2b connectivity for α data set, 816 kbps network, 2 W tx power



A cursory examination of the connectivity of various nodes in Figure 4.2a sheds some light on the true connectivity situation. Most of the nodes in the network require an average of between two and three hops to reach most other points in the network. This figure is higher for nodes located on the network's periphery, such as nodes 1 and 4. Node 1 requires an average of at least four hops to reach most other nodes in this network configuration.

Let us assume that the following video VC links are required: nodes 24 - 8, nodes 2 –17, and nodes 7 – 10. This would require ten 32 kbps slots, and this is not available for the bit rate used. Other link combinations would be supportable (e.g. nodes 11-23, nodes14 – 1, and nodes 10-12), but in many cases the required connectivity will not be achievable.

The 816 kbps data rate would permit longer paths to be supported, but as Figure 4.2c demonstrates, the connectivity map is significantly eroded from the previous case. With only 11% of the possible links available, sufficient connectivity is simply not achieved to form an all-encompassing network.

In addition to the problems raised by these configurations, the requirement for some form of centralized or hierarchical demand access control was basically ignored, although the control channel data rate was raised. It is difficult to envisage a centrally controlled architecture sufficient to provide for efficient scheduling in such a band-limited environment. This also conflicts with the requirement to have a system operating in a survivable manner.

There is little more that could be done to permit a TDMA system to effectively handle this problem. Transmit power could be increased to provide a corresponding increase in the cut-off threshold. However, in view of the performance of the 2 W cases, it is evident that to make meaningful improvements, a dramatic increase in power would be required. At a minimum this would impact on the mobility and portability of such a system.

Additional transmit channels at different frequencies also could be added to provide what would effectively be a FDMA/TDMA structure. This would have several implications. First, additional transceivers would be required to allow for simultaneous channel operation at each terminal, and some interference from one system to another could be expected. Second, this would also impact on power requirements for the additional transmitter(s).

If five TDMA channels were used with the 328 kbps system, and the power for each channel was set at 0.4 W, poor overall connectivity still results. The low cut-off threshold would not yield a workable system. If the data rate for each of these channels was further reduced to 70 kbps (still providing 328 kbps), then the cut-off threshold remains low (133 dB) and therefore once more sufficient connectivity is not achieved.

Even if connectivity under these conditions was acceptable, it would be extremely difficult with these data rates to provide the links required to support the specified network requirements of four voice circuits, three video circuits and the various data circuits.

For these reasons, although TDMA continues to be a key wireless technology, it does not provide an acceptable network solution for the integrated multimedia characteristics as defined in Chapter 1. Accordingly, pure TDMA and TDMA/FDMA architectures will not be developed further for this network.

In order to provide a workable network capable of supporting the various voice, video and data services in a distributed wireless environment, some other multiple access technique will have to be investigated. Accordingly, spread spectrum technology is next examined for application to this system.

4.3 CDMA

Spread spectrum technologies have been in use for several decades in a wide range of applications. Initially, spread spectrum systems were focussed on military technology where low probability of intercept (LPI) or anti-jamming capabilities were a critical consideration. Its application has now spread to widely available commercial systems due to the ability of this technology to operate in the presence of interference.

Spread spectrum technology takes two basic forms: frequency hop spread spectrum and code division multiple access. The basic concept of spread spectrum involves spreading the intended message while transmitting, and despreading at the receiver while simultaneously spreading the added narrow band noise/interference. In frequency hop systems the carrier frequency is not constant. Frequency hop systems are not discussed here due to the added complications arising to the lack of a constant transmit band and the relative elegance of the code division approaches.

4.3.1 CDMA Techniques

Code Division Multiple Access (CDMA) techniques are primarily based on direct-sequence spread spectrum (DSSS). CDMA is increasingly quoted in wireless literature and is also to be found as a keystone technology in many second and third-generation wireless personal communication systems. Indeed CDMA has been adopted by the Telecommunication Industry Association as the TIA/EIA IS-95 standard for cellular and by the Alliance for Telecommunications Industry Solutions as J-STD-008 for PCS.

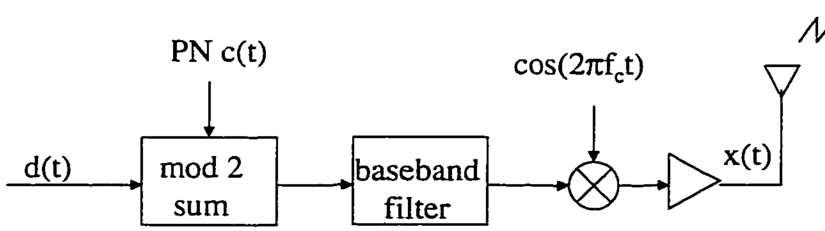


Figure 4.3: Basic CDMA system transmitter [4]

As illustrated in Figure 4.3 above, in CDMA signals the spreading is accomplished via a pseudo noise (PN) sequence used at the transmitter. Despreading is accomplished by applying the same PN sequence at the receiver. The basic concept of CDMA is as follows, using coherent binary phase shift keying (BPSK) for illustrative purposes. BPSK is the simplest form of such a communications system, although quadrature phase-shift keying (QPSK) is a more common method.

The encoded CDMA BPSK signal is given by

$$x(t) = c(t)s(t) = c(t)d(t)\sqrt{(2S)}\cos(w_c t)$$

where $d(t)$ = the baseband signal at the transmitter input
$c(t)$ = the spreading signal
S = signal power
w_c = carrier frequency

The signal $s(t)$ has a $[(\sin x)/x]^2$ spectrum with approximate bandwidth of $1/T$ and the spreading signal has bandwidth $1/T_c$. The processing gain of this system is thus equal to T/T_c. If a narrow band noise $n(t)$ is added, then at the receiver we receive $x(t) + n(t)$. This received signal is multiplied by the pseudo-noise waveform $c(t)$. As $c^2(t) = 1$, we then have $x(t) + c(t)n(t)$, thus the noise is effectively spread and therefore attenuated while the intended message is not.

CDMA offers many key advantages for the application under investigation here. Of perhaps the greatest significance is CDMA's ability to flexibly support many simultaneous transmissions throughout the network without the requirement for frequency plans and channel allocation, and without experiencing a significant scheduling problem. This is because simultaneous transmissions, provided they are intended for different destinations, appear as interference that is attenuated by the processing gain of the spreading sequence.

In a TDMA system, only one transmitter can be active at a time, otherwise zero capture dictates that signals will destructively interfere with each other. In a CDMA system, several transmitters can be simultaneously active. However, although many transmissions can coexist (again dependent upon their interference on each other), a given conventional receiver can only receive one signal at a time.

The limitation with TDMA is bandwidth and power. In CDMA, the limitation is, instead, interference. CDMA systems will experience graceful degradation in overall performance as the number of interferers increases. However, because multiple interferers are involved, there is a reduction in the fluctuation of the total interference power caused by long and short term fading.

Signal capture is accomplished through the use of unique codes which permit true simultaneous multiple access. Collisions are of no real consequence unless the noise floor has been saturated by too many concurrent signals or two packets are simultaneously attempting to connect to the same destination receiver. CDMA combats multipath interference, and introduces narrow band interference rejection. It should be noted that this anti-jamming capability is still a key consideration for military systems that must operate in an electronic warfare environment. Furthermore, as the transmitted signals are spread over a wide bandwidth, signal detection, interception and ranging is made much more difficult.

The ability of CDMA to provide low probability of intercept (LPI) or direction-finding also remains of particular interest to military communicators. In hostile environments, tactical stations must not give away precise locations to hostile direct-finding (DF) equipment, because modern DF techniques combined with artillery can be extremely lethal.

Furthermore, the lower power consumption of CDMA terminals is advantageous for mobile networks. In the case of cellular systems, CDMA mobiles can function with less than one tenth of the peak power for a TDMA phone operating at a comparable data rate. CDMA can actually use multipath fading advantageously by combining multipath signals to produce an enhanced signal via the use of RAKE receivers and other signal processing techniques [5].

4.3.2 Near-Far Problem

A major complication that can be encountered with CDMA systems is the so-called near far problem. This is illustrated in Figure 4.4 below.

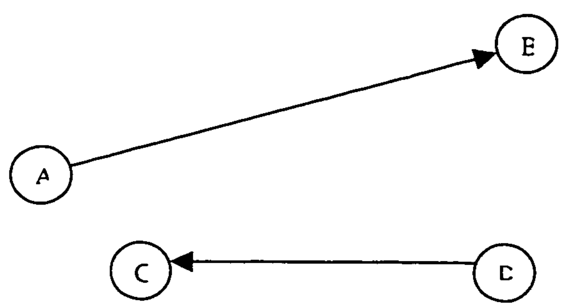


Figure 4.4: Near-Far Problem

In Figure 4.4, nodes A and D are simultaneously transmitting via CDMA to their target nodes B and C. If no power control is in effect, and node C is much closer to A than node D, then the near-far problem will result. Although node C will see node A's transmission in an attenuated form (the power transmitted by A, minus the path loss, minus the processing gain of the spreading factor), it can still overwhelm the signal from node D if the path loss from D to C is relatively large. Power control is normally used in CDMA systems to mitigate the effect of the near-far problem, but its implementation can be difficult to accomplish in a distributed setting. This will be discussed in greater detail in subsequent chapters.

4.3.3 CDMA Code Assignment

CDMA codes can be assigned in a variety of ways. A receiver-oriented spread spectrum transmission protocol employs a 'unique' pseudo noise sequence for each receiver. It should be noted that the term 'unique' is used loosely as codes can be reused throughout a network if transmission range sufficiently attenuates signal strength. This is analogous to frequency reuse in cellular networks. The use of unique codes can be very problematic in large commercial systems where it may be impossible to have sufficient unique codes to support the system. However, given the small number of terminals in this network this is not viewed as a significant constraint.

In a receiver-oriented protocol the transmitters must be pre-loaded with all required receiver codes and make a suitable selection based upon the intended receiver (or the first receiver in the transmit path). Similarly, a transmission-oriented protocol assigns unique codes to the transmitter. In this case the receivers must scan a series of codes and lock on when one is received, unless some form of scheduling is used that would allow receivers to look for specific codes at specific times.

It is also possible to utilize a protocol in which a common spreading sequence is used for all transmissions. In this case the discrimination of incoming packets must be based upon the time of arrival for the packets in question and time offsets. This protocol does not provide for good signal capture and can result in interference between packets transmitted to different receivers. This does however facilitate the transmission of broadcast messages.

Lin and Gerla recommend the use of transmitter oriented protocols. One of the key arguments used in selecting this protocol is that "receiver based assignment cannot avoid intercluster collisions [13]." However, this is a simplistic approach that assumes the overarching use of a clustering protocol. It does not hold true if one makes use of CDMA and creative scheduling techniques or channel sensing schemes such as Busy-Tone Multiple Access (BTMA) [14]. BTMA simply introduces a busy tone that a node transmits if it is locked receiving another station's transmission. It offers one possible solution to the contention problem and will be examined in greater detail in the following chapters.

Scheduling options afforded by the architecture developed in Chapters 4 and 5 can also mitigate collisions to a large extent. For example, the introduction of a variable slotted-delay for back-off can reduce future contention of colliding call attempts (these concepts will be further developed in Section 5.5.

The question of code orientation is critical when examining CDMA designs and determining how packets will be distributed in the network. Will the code be transmitter oriented, or receiver oriented, and will some form of broadcast or flooding code be employed? These broadcast methodologies will be explored in greater detail in Section 6.1.

4.3.4 System Requirements Applied to CDMA

The initial demonstration of CDMA capabilities in this case will provide examples of pure, asynchronous CDMA and then a synchronous CDMA system with simple time slots. The latter is done to allow for some measure of QoS guarantee provision, as required by VC-based services. However, a very basic CDMA example with no such assurances will begin the discussion.

The pure CDMA and slotted CDMA examples will utilize one basic assumption: control channel traffic will be carried on a separate 16 kbps TDMA channel. This is because it is necessary to allocate a unique time slice to each terminal for control purposes. This cannot be done on an asynchronous CDMA basis, because there is no guarantee that one-hop nodes would be in a position to receive the signal, and delivery of the connectivity information must be guaranteed. Thus, only the carrier services will be examined.

In this case, the Eb/No calculations allow for us to utilize the effective data transmission rate rather than the overall actual system transmission rate. However, in view of the requirement to allow for simultaneous transmission, we must take into account the effect of these interferers.

One formula in wide use for calculating the capacity of a CDMA system with multiple interferers is: $M < (B/R)/(Eb/No_{\text{required}})$, where M is the maximum number of users and B/R is the processing gain. This calculation will not be used here because it assumes the implementation of a perfect power control system that equalizes signal reception at each node. In view of the difficulties associated with achieving this in a distributed network, a more realistic interference model must be developed.

A simple distribution of emitters must be devised to provide a basic simulation of the interference terminals must work through, thus leading to estimates of network connectivity. This begins with the assumption that four voice, three video, two low speed data and two medium speed data links are simultaneously active, in accordance with the system specifications listed in Section 1.3. The channel model output for the α data set reveals that the adjusted path loss between points ranges from extremes of 94 to 169 dB. The vast majority of the points are found in the range of 135 to 155 dB and the average path loss for all links is 144.8 dB.

Any discussion of link budgets will have to include consideration of these interferers. To provide a reasonable interference model, we assume that:

- the voice transmitters are located such that path loss from these points to the intended receiver is 130, 140, 150 and 160 dB;
- the video transmitters are located such that path loss from these points to the receiver is 130, 140 and 150 dB;
- the low speed data transmitters are 135 and 145 dB away from the receiver; and,
- the medium speed data transmitters are 135 and 145 dB away from the receiver.

The distribution of points for the α data set indicates that these assumptions are reasonable. Of the 300 adjusted path loss figures, approximately 10% have a path loss of less than 130 dB, and the majority have a path loss within the 135 to 155 dB range. Therefore, these assumptions can be viewed as a conservative model of the effect of interferers.

Other assumptions that are used include a common transmitter power of four different levels (0.1 W, 0.4 W, 1W, and 2 W as for the TDMA cases), a chip rate of 10^7 chips/sec, a noise temperature of 2000°K and a 10 dB signal to noise margin required. If the above assumptions and figures are used, it is possible to determine the total noise power including interference from the emitters, and therefore threshold cut-off via the following formula:

$$\frac{20 \text{ dBm} - \text{path loss (of intended path)}}{R\left(No + \sum^{i}\left([20 \text{ dBm} - \text{path loss (of interferer } i \text{ to receiver)}]/B\right)\right)} \geq 10 \text{ dB}$$

These calculations have been performed for the four data types, transmitted at the various power levels, and are summarized in the following section. It is evident that the most sensitive channel is the one provided for video use. This is due to the much higher data rate utilized by this channel.

The pure CDMA approach is very limited in one key respect: each terminal can only carry one type of circuit at a time. In a packet radio network where connectivity is relatively low, this will not support the specifications that were provided in Chapter 1. If the threshold figures were higher, this would not be a concern because each node could be capable of contacting each and every other node, and multi hop paths would not be a consideration. However, multi-hop paths are important, and if each node can only handle one circuit at a time, then very few connections will be possible. For this reason, a simple-slotted CDMA system will also be investigated.

4.3.5 Simple-Slotted CDMA

In a network where connectivity is not 100% (i.e. not every node can contact every other node), the requirement to carry different data types through a packet radio network leads to a requirement for individual terminals to be capable of carrying different services simultaneously. In its most basic form this leads to a CDMA system with the addition of TDMA time slots to permit the quasi 'simultaneous' provision of different services through the same point. This time slicing introduces the need for network synchronization.

The most basic manifestation of such a system for the services defined in Chapter 1 would include provision of each data type through each terminal. This would require a data rate of 50 kbps. This includes 32 kbps video, 8 kbps voice, 2 kbps LSD and 8 kbps MSD. For the purposes of this research this architecture is termed simple-slotted CDMA.

The ramifications of making such a modification to a CDMA system are threefold. First, the data rate used in the link budget calculation is increased, thus reducing the carrier to noise margin. Second, the processing gain of the spreading function is also decreased, leading to a further reduction in the achievable margin. Although these first two affects lead to performance degradation when compared to pure CDMA, the final effect does the opposite. The slotting/synchronization reduces the number of signals that can interfere with each other because, for example, only video circuits can be transmitted at the same time, and voice circuits will not interfere with datagram transmissions. Because CDMA systems are generally interference limited, this can lead to an overall performance improvement.

The link budget for such a system has been calculated and is also summarized in Figure 4.5 below.

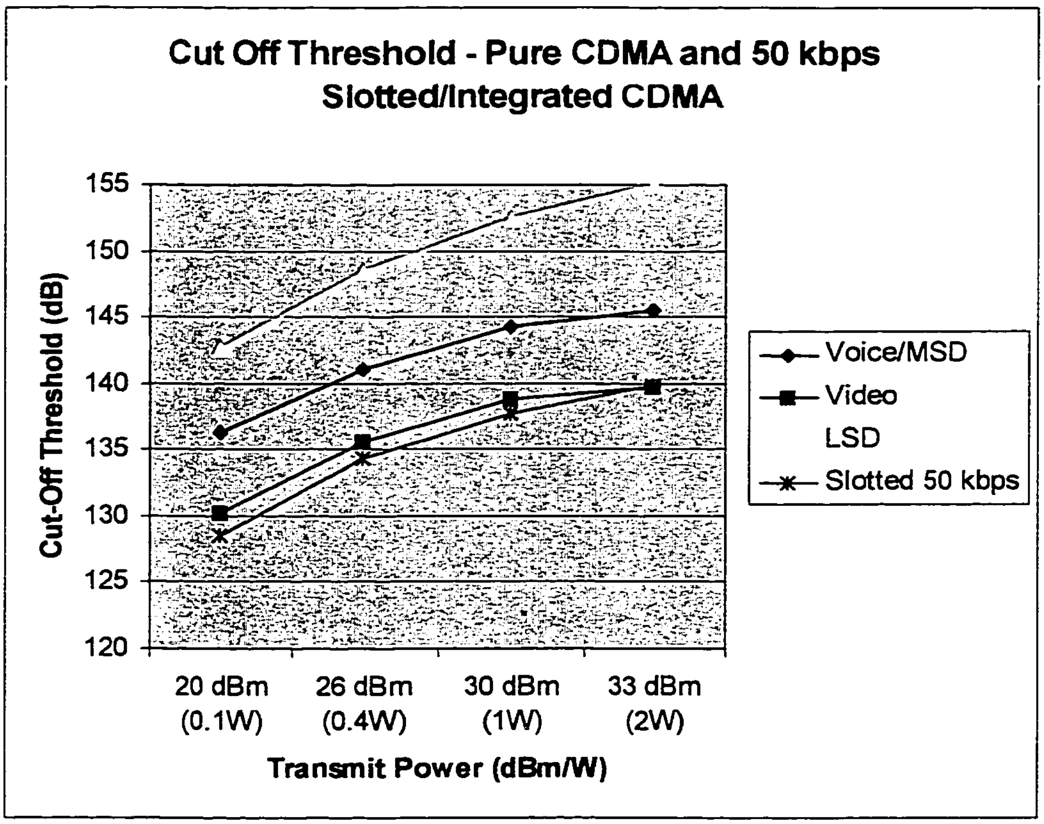


Figure 4.5: Cut-off thresholds: integrated services on CDMA and simple, slotted CDMA

4.3.6 Comparison – TDMA and CDMA

In general, it can be concluded that the CDMA examples provide much better connectivity than the TDMA cases, as evidenced by the significantly higher cut-off thresholds depicted in Figures 4.1 and 4.5. These higher cut-off thresholds directly translate into enhanced connectivity.

Care must be taken when comparing the data rates of the two different multiple access techniques. For example, the data rate used for the simple-slotted CDMA example was 50 kbps as opposed to 328 kbps for the least capable TDMA example. However, the 50 kbps raw data rate figure represents the peak capacity of each of the 25 nodes rather than the aggregate. Therefore the CDMA example provides a higher potential aggregate data rate than for the TDMA case.

As mentioned, the CDMA examples provide significantly better connectivity than in the TDMA cases. Figure 4.6 provides the connectivity map for the 2 W 50 kbps slotted CDMA example, for the α data set. In this example, 103 connections are possible between the various nodes. As a result, the vast majority of node-to-node connections are just one or two hops away. A small number of connections require three hops, but no connections require four or more. This contrasts most favorably with the TDMA examples.

Therefore, in view of CDMA's better connectivity and higher data rates for the same transmit power, it is determined that it is the superior multiple access technology for this network design. Accordingly, TDMA is omitted from further analysis for this system solution.

4.3.7 Comparison – Pure CDMA and Slotted CDMA

The effect of decreasing the number of signals that can interfere with each other can be seen by comparing the CDMA – video, and slotted CDMA cases. For the lower transmit powers, the higher number of interferers in the video example does not impact as heavily as the higher data rate for the slotted system. However, as the power increases, the effect of the larger number of interferers in the video case becomes more significant.

At 2 W the cut-off threshold for the simple-slotted example surpasses the video result even though the video links are operating at a lower data rate. Thus, some efficiencies can be gained by using a slotted methodology in combination with CDMA The connectivity of the 2 W 50 kbps slotted CDMA example is shown in Figure 4.6. This map indicates that all points in the network can be reached by any other point in at most three hops, and in most cases only one or two hops will be required.

Of added significance is the enhanced ability of slotted CDMA to support multi-circuit operation at each node. The pure CDMA system would provide an acceptable solution if each node was only one hop away from all other nodes. However, the cut-off threshold for the video CDMA is approximately 139 dB (Figure 4.6 provides an acceptable approximation).

For such a sustainable path loss, nodes in the network would require one to three hops to reach any intended destination. Thus pure CDMA will generally not support the system's stated requirements to simultaneously carry three video, four voice, two low speed data and two medium speed data connections. With twenty-two nodes active, it is inevitable that some links (i.e. any of the two or three hop links) will require intermediate nodes. An intermediate node is not capable of initiating or receiving its own information while it is dedicated to maintaining another link, or its own communications.

Therefore, although the slotted system degrades the overall ability to provide lower rate services such as voice and low speed data, the flexibility of each terminal is dramatically increased, and it provides a more integrated approach to the problem of multimedia service provision. This justifies further examination of the slotted CDMA technique.

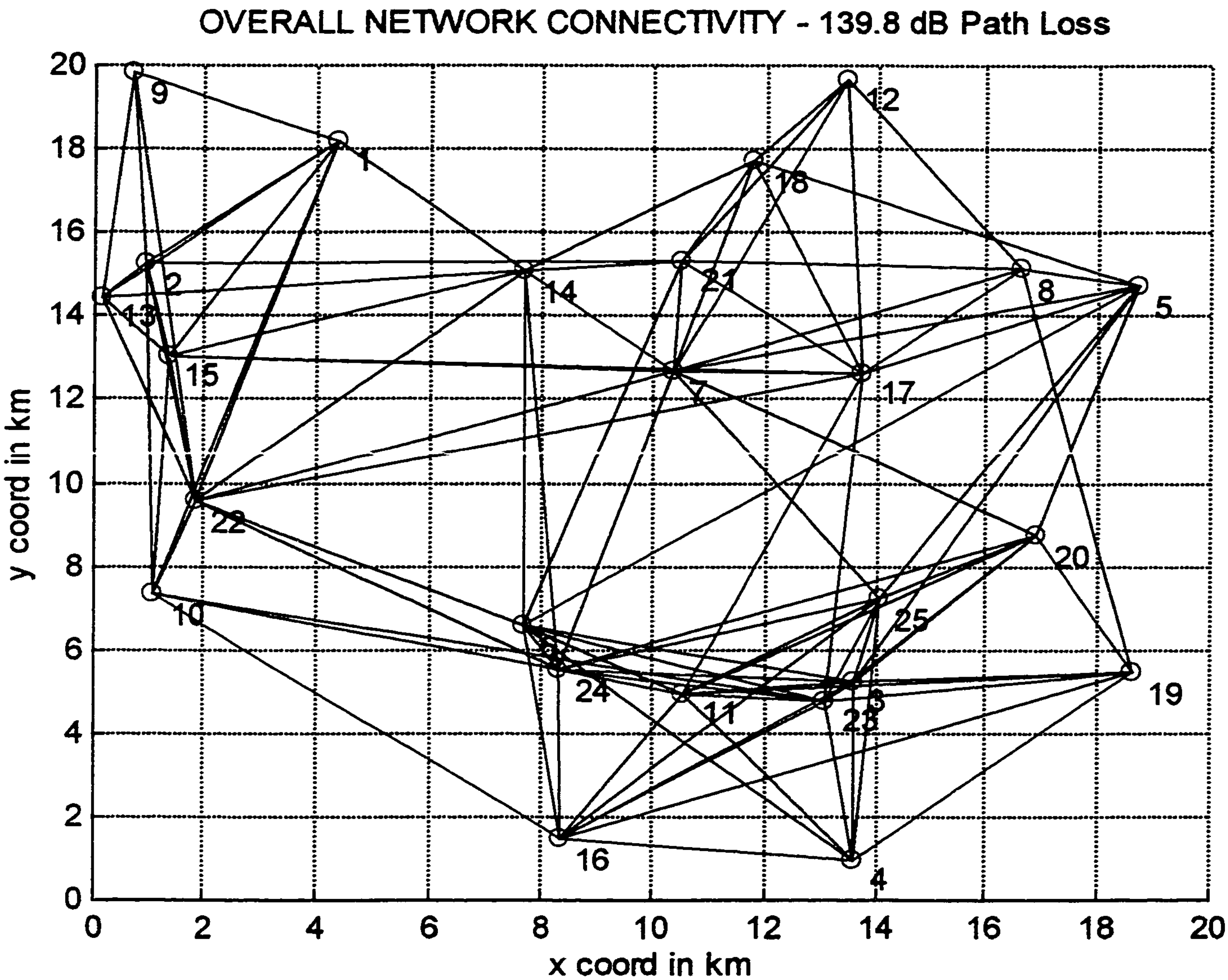


Figure 4.6: Connectivity for α data set, 50 kbps slotted CDMA, 2 W tx pwr

4.4 Slotted CDMA for Multimedia Packet Radio

It has been shown that CDMA offers better performance than TDMA for the specifications defined in Chapter 1. It was also demonstrated that pure CDMA approaches also require enhancement to achieve these same specifications. As observed in the previous section, when attempting to achieve an acceptable trade-off between efficiency and flexibility, a slotted CDMA approach can become more appealing.

A basic slotted CDMA architecture was used to illustrate how multiple circuits can be provided through individual nodes, and that at higher transmit powers this can actually result in a performance improvement by reducing the number of simultaneous interferers. This maintains the advantages of CDMA, while permitting us to provide QoS for multiple VCs and connectionless data circuits.

Therefore, a slotted CDMA methodology will be investigated for application in this system. This must include an examination of several issues, including the provision of a control channel, definition of the specific data structure, routing and scheduling for the network. The degree of slotting actually required must also be examined. These topics and others will be pursued in Chapter 5.

5.0 Network Control for Slotted CDMA: Data Structure, Routing and Scheduling

5.1 Network Data Structure

In Chapter 4 it was determined that TDMA was not a suitable multiple access technique for this network design. It was also determined that some form of a slotted CDMA structure would be more appropriate than a pure CDMA architecture for the specifications presented in Chapter 1. Thus, slotted CDMA will be probed in greater detail to reach a more complete framework for the network.

A detailed investigation of the options available for slotted CDMA must begin with a closer look at the network data structures that could be available. Different degrees of slotting can be achieved, although as was shown in the previous chapter, increasing the data rate can decrease the path loss threshold.

Once the data structures for investigation have been determined, they will be compared on the basis of connectivity and QoS provision. The selected structure will then be examined in terms of routing and scheduling, both critical issues requiring attention in a packet radio network. The construction of routing tables at source nodes will also be considered in tandem with the control channel specification, and finally scheduling and reservation will be addressed to ensure that QoS guarantees can be met for the various types of services required in the system specification.

5.1.1 Cell Size Assumption

One assumption that will be made before the comparison and examination of the options is that any data carried in the network will be encapsulated in a pseudo wireless ATM (WATM) structure similar to that proposed by Raychaudhuri [15]. This structure modified the standard ATM cell from 53 to 55 bytes in length to accommodate the additional error control coding required for porting ATM from a fiber environment to the wireless world. Thus, for a 48 byte of payload data, 7 bytes of overhead is required. This ratio of 384 bits/440 bits translates into a requirement for an additional 15% of data rate capacity.

5.1.2 Network Control Channel

The network control channel is critical in this distributed design. Without the control channel, nodes in the network would be incapable of making routing decisions. However, although there is the clear need to promulgate connectivity throughout the network, there is also the simultaneous requirement to do so in a highly efficient manner. Bandwidth available for the control channels is limited, as opposed to the 'wired' world where the system configuration is virtually static and bandwidth availability is rarely an issue.

The CDMA examples provided in Chapter 4 assumed the provision of separate TDMA control channels. At first glance, it appears that this would not be of great concern. However, it does introduce key disadvantages. First, it requires the provision of separate transmitters and receivers, thus increasing power consumption, complexity and cost. In a less benign environment, such a TDMA channel would be susceptible to jamming, and would also eliminate the LPI characteristics of a CDMA solution.

In view of these considerations, a stand-alone TDMA control channel does not survive scrutiny. Therefore, the slotted CDMA options pursued in this section will include capacity for the control channel as defined below. This capacity will have to be integrated as a completely separate slot as no two nodes should be transmitting control information at the same time.

A key consideration for operating a network in a distributed environment is the ability of terminals to learn of their own connectivity, and that of other nodes in the network. As mentioned, without such a mechanism it would be impossible to route packets. To accomplish this 'learning', there is an initial requirement build node connectivity tables at each station, and this information is based on the connectivity data passed in the control channel.

The network's configuration is dynamic because nodes can move and propagation conditions can change. Therefore the control channel must also support the ongoing requirement to maintain and update connectivity and routing tables at each station, thus each node must update and retransmit its own connectivity table on a periodic basis. The connectivity tables should include information regarding one hop neighbors, and information detailing the connections available throughout the network. The routing tables will be built from the connectivity tables and will be discussed later in this chapter.

The process of building the connectivity tables begins at system start-up where the system must learn its own topology. On initialization, the nodes begin to 'ping' in a predetermined sequence according to their node ID, each of which is unique, numbered from 1 to 25. Initially, the ping merely indicates the node's identification, essentially transmitting "I am node X" so that other nodes in range can learn of its presence. After the other nodes have done their initial ping, subsequent repetitions require each node to transmit a connectivity table indicating who they hear, who their one hop neighbors hear and so on.

For very large networks such a mechanism would not be feasible. This would likely lead to network segmentation, possibly through clusterheads as already discussed. However, in our case the network consists of a relatively small number of stations and therefore this process can be done directly.

It was initially estimated that 640 bps of control capacity was required for each node in the network. Integration of the control channel with 48 byte/55 byte WATM cells therefore requires the allocation of two cells for each node in the network. Thus, inclusion of the control channel will add a requirement of 22 kbps to the network's capacity (25 nodes x 2 cells/node per sec x 55 bytes/sec x 8), and the actual control channel capacity at each node becomes 768 bps(2 x 48 x 8).

The control channel must include several components to allow the terminal to build connectivity tables and then routing tables. For each station, this will include:

- its ID number (1 byte);

- the identity of its one hop neighbors via a 25 bit binary array with a 1 indicating the existence of a link; and,

- the known connectivity of all other nodes in the network through a binary 25 by 25 bit matrix of 625 bits or 79 bytes, again with a 1 in the I-J element of the matrix indicating the existence of a link between nodes I and J.

It should be noted that the latter element could also be implemented by the transmission of link-state changes. However, this would require the transmission of considerably more data than a simple 625 bit table, unless no changes occurred. This is due to the fact that a unique node identifier would have to be used (consisting of one byte each), thus each change would be represented by two bytes rather than one bit. Therefore, in view of the bandlimited environment, the simple connectivity table will be used.

5.1.3 Control Channel Structure

Given that two control cells per node are allowed per second, and that each cell consists of 55 bytes with a 48 byte payload, 96 bytes are available per second per node for control. The following is proposed:

First control cell: (per second)	1 byte - node ID 4 bytes - one hop neighbors 40 bytes - first half of the 25x25 node known connectivity matrix 3 bytes - spare
Second control cell: (per second)	1 byte - node ID 4 bytes - one hop neighbors 39 bytes - for second half of known connectivity matrix 4 bytes - spare

It should be noted that the node ID and one-hop neighbors are repeated in each cell (i.e. twice per second) and that several spare bytes are available for future use.

5.1.4 Network Synchronization

The selection of a slotted CDMA methodology requires the imposition of a robust, highly synchronized network structure. In the absence of a fixed infrastructure and a central control node, how can this be accomplished? The answer lies with a Global Positioning System (GPS) based solution.

GPS is already in widespread use on the battlefield for positional location with a high degree of accuracy. One of the added benefits of GPS is the global clock, based on the atomic clocks integral to each GPS satellite. This method will allow for synchronization throughout the network without a high degree of complexity. The addition of GPS technology to the network transceivers will not result in prohibitive cost increases. Indeed, basic receivers are now available to consumers for less than $200.

5.1.5 Options for Slotted CDMA

The range of slotted CDMA options available for examination is wide, ranging from low capacity solutions to solutions providing high capacity at each node. Three slotted CDMA options in this range will be evaluated in this section for consideration in the network design. Each option includes its own data structure.

The first option will consist of the 50 kbps example used in Chapter 4, with the addition of the control channel identified above and WATM encapsulation. It will be termed simple-slotted CDMA, and permits each node to carry multiple circuits, although only one channel of each type could be supported simultaneously. Thus one node could, if necessary carry one speech, one video, one LSD and one MSD circuit in addition to its control traffic.

Second, an example with approximately twice the aggregate data rate will be considered. This is termed mid-slotted CDMA. It will permit a second voice and video circuit to flow through each point since these are more common than the datagram circuits and are required to adhere to more stringent delay characteristics.

Finally, a high capacity option termed max-slotted CDMA, will be examined. This option will have sufficient capacity to support a worst case scenario where all traffic (148 kbps effective rate) must be routed through a single point. Figure 5.1 below provides a stylized illustration of this case, which would occur if the network was fractured in two distinct groupings, with only one point of connection between them. If all the traffic originated from points on one side to points on the other, such a capacity would be required. The last two options also both include the use of control channels and WATM cells.

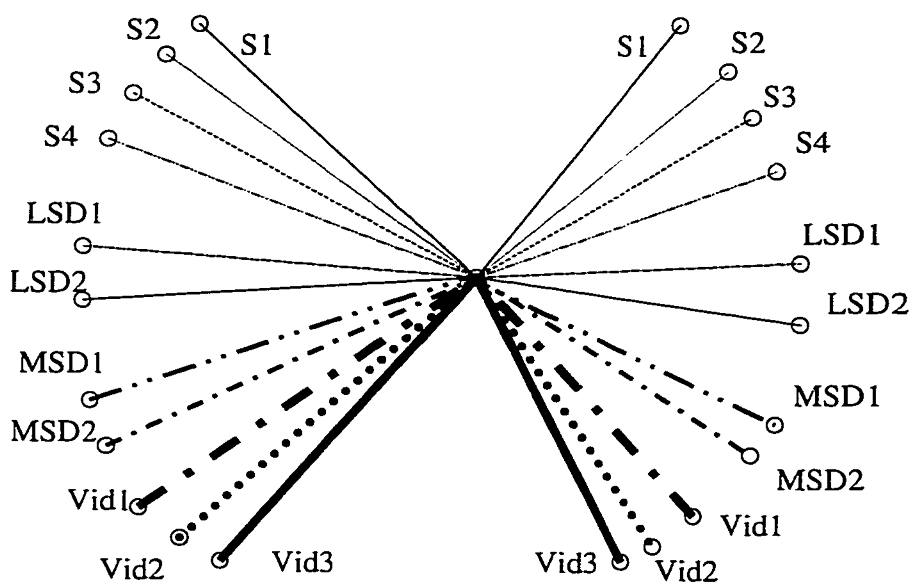


Figure 5.1: Worst case capacity for single node

For each of the three cases a new link budget will be calculated and connectivity diagrams will be produced. Table 5.1 below summarizes the data rates required to support these configurations, based upon the data rates previously specified, addition of a control channel and the use of WATM cells.

Service	**Simple-Slotted**	**Mid-Slotted**	**Max-Slotted**
Low speed data	2kbps = 5.2 cells/s → 6 cells/s	2kbps = 5.2 cells/s → 6 cells/s	2kbps = 5.2 cells/s → 6 cells/s x 2 → 12 cells/s
Medium speed data	8 kbps = 20.8 cells/s → 21 cells/s	8 kbps = 20.8 cells/s → 21 cells/s	8 kbps = 20.8 cells/s → 21 cells/s x 2 → 42 cells/s
Video	32 kbps = 83.3 cells/s → 84 cells/s (42 in and 42 out)	32 kbps = 83.3 cells/s → 84 cells/s x 2 (42 in and 42 out) → 168 cells/s	32 kbps = 83.3 cells/s → 84 cells/s x 3 (42 in and 42 out) → 292 cells/s
Voice	8 kbps = 20.8 cells/s → 22 cells/s (11 in and 11 out)	8 kbps = 20.8 cells/s → 22 cells/s x 2 (11 in and 11 out) → 44 cells/s	8 kbps = 20.8 cells/s → 22 cells/s x 4 (11 in and 11 out) → 88 cells/s
Control	22 kbps → 2 cells/s per node → 50 cells/s	22 kbps → 2 cells/s per node → 50 cells/s	22 kbps → 2 cells/s per node → 50 cells/s
Total Bit Rate (actual)	183 cells/s ⇒ 80.5 kbps	289 cells/s ⇒ 127.1 kbps	484 cells/s ⇒ 213.0 kbps
Processing Gain $Mc = 1.5 \times 10^7$	22.7 dB	20.7 dB	18.5 dB

Table 5.1: Comparative data rates – simple-slotted, mid-slotted and max-slotted schemes

The application of the data from Table 5.1 in the link budgets will be done similarly to the method utilized in Chapter 4, with one exception. Only the 2 W transmit power will be used as it provides the best connectivity.

Although the interference model has been kept simple, the use of multi-hop paths could complicate this issue as different points on the same path may transmit at the same time. For simplicity, this is ignored for the moment. Specific examples will be used to determine if this is a bona fide concern.

In these cases it is useful to observe the link budget results for individual portions of the slotted structure. For example, the control channel will not result in any interferers, and the likelihood of interferers in the datagram circuits is lower because fewer circuits are active. Also, the control channel and datagram channels will in reality require more reliable delivery than the information transmitted for the voice and video channels. While voice and video traffic is delay sensitive, these VC services can operate under lower Eb/No conditions. Table 5.2 summarizes the two different situations for the three cases.

	Simple-Slotted	**Mid-Slotted**	**Max-Slotted**
Video portion	137.7 dB	135.7 dB	133.5 dB
Control channel (no interferers)	139.5 dB	137.5 dB	135.3 dB

Table 5.2: Cut-off threshold for 10 dB margin

5.1.6 Eb/No Requirements

Up to this point, only one number has been used for the acceptable Eb/No, namely 10 dB. One possibility at this stage is to correlate the data types (voice, video, data, and control) with more the appropriate Eb/No figures. For these purposes, the calls are categorized as error sensitive and non-error sensitive.

Clearly, the low speed data, medium speed data and control channels are error sensitive whereas the voice and video are not. That is not to say that errors are relatively unimportant in the voice and video realm. While errors in these types of calls will be a nuisance, they will not be disruptive in nature. This is not the case for the data and control channels. A corrupted control channel could disrupt the entire system, rendering it unusable.

For our baseline calculations, the Eb/No requirements for the two traffic types are set as follows in Table 5.3.

Traffic Type	**Bit Error Rate**	**Eb/No** (for BPSK)
Error Sensitive	10^{-5}	9.6 dB
Non-Error Sensitive	10^{-3}	6.8 dB

Table 5.3: Bit error rates and Eb/No requirements

It should be noted that these numbers apply to the case of binary phase shift keying (BPSK) which will be assumed throughout in this case. The error sensitive case (bit error rate = 10^{-5}) may appear to be unusually permissive, particularly in comparison to a fiber environment BER of 10^{-9}. However, the provision of additional coding could further decrease this rate and there will be sufficient resources available to accomplish this through the additional overhead furnished by the WATM cells.

The information in Table 5.2 is specified for video and control traffic, but can be more generally described as non-error sensitive and error sensitive. The error sensitive traffic is essentially interferer free, while the non-sensitive portion uses the same interferer model, data rate and processing gain. Since error sensitive traffic requires an Eb/No of 9.6 dB, we can replace the earlier 10 dB margin with this figure. Similarly, we can replace the same 10 dB margin for the VC traffic with 6.8 dB. This results in Table 5.4.

	Simple-Slotted	**Mid-Slotted**	**Max-Slotted**
VC traffic (not error sensitive)	140.9 dB	138.9 dB	136.7 dB
Data traffic (error sensitive)	139.9 dB	137.9 dB	135.7 dB

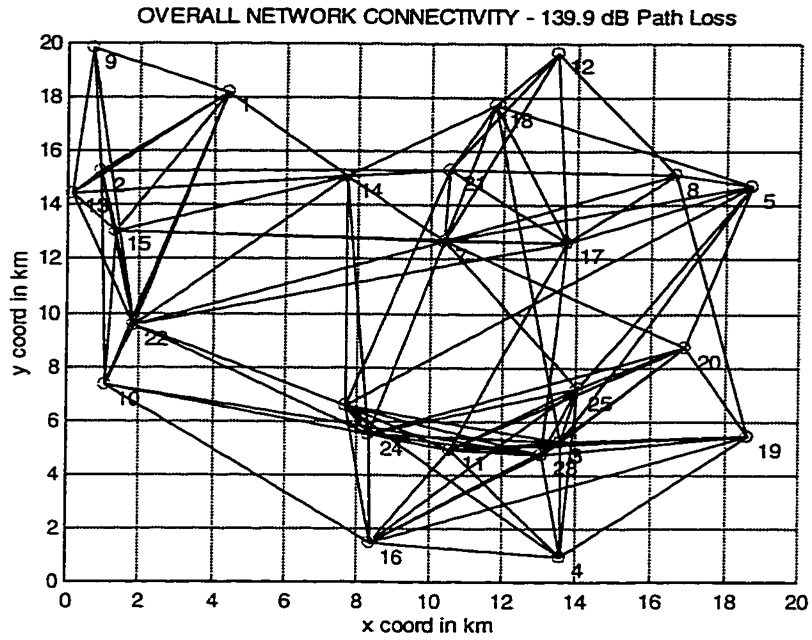
Table 5.4: Cut-off threshold for 6.8 dB/9.6 dB Eb/No

For each of the three columns in Table 5.4, the lower figure is selected to determine the sustainable path loss threshold for the overall network. These cut-off thresholds have been used in conjunction with the channel model to provide a simulation of the network connectivity for the different three cases.

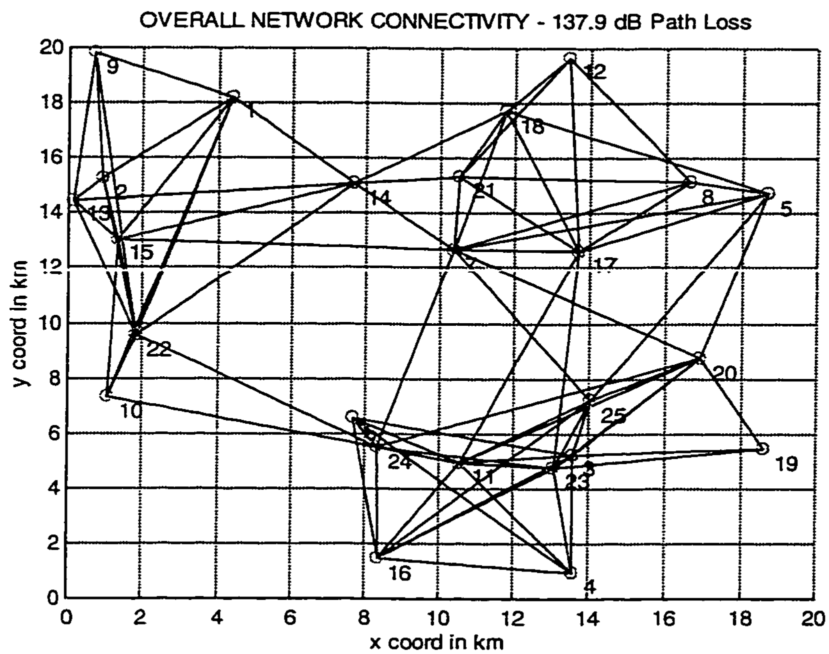
Utilizing the various cut-off thresholds, the resulting network connectivity maps are illustrated in Figures 5.2, 5.3 and 5.4 on the following pages for the three slotted CDMA options and three of the five data sets.

The network connectivity diagrams provide a graphical comparison of the connectivity provided by each mechanism. Observations regarding these connectivity maps will follow.

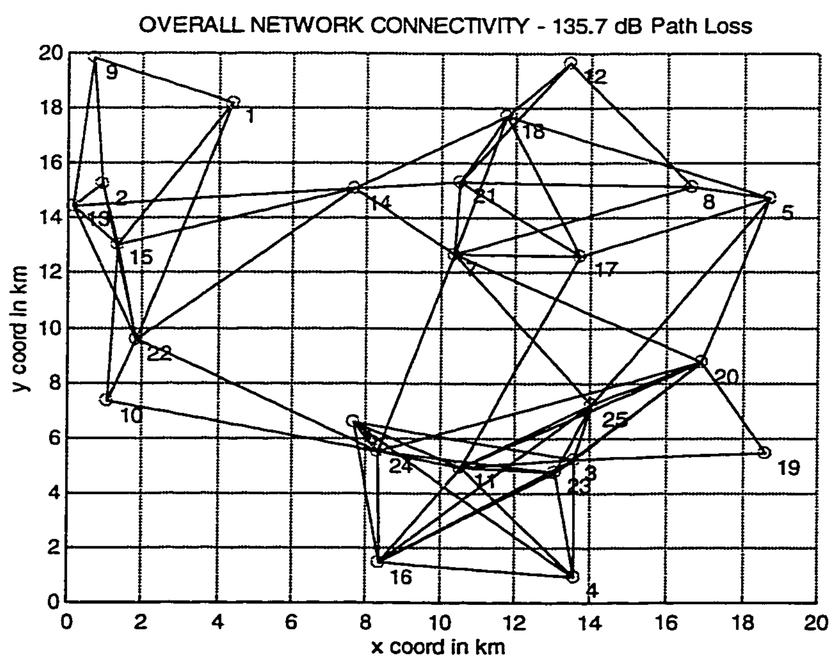
5.2a overall network connectivity
139.9 dB sustainable path loss
α data set



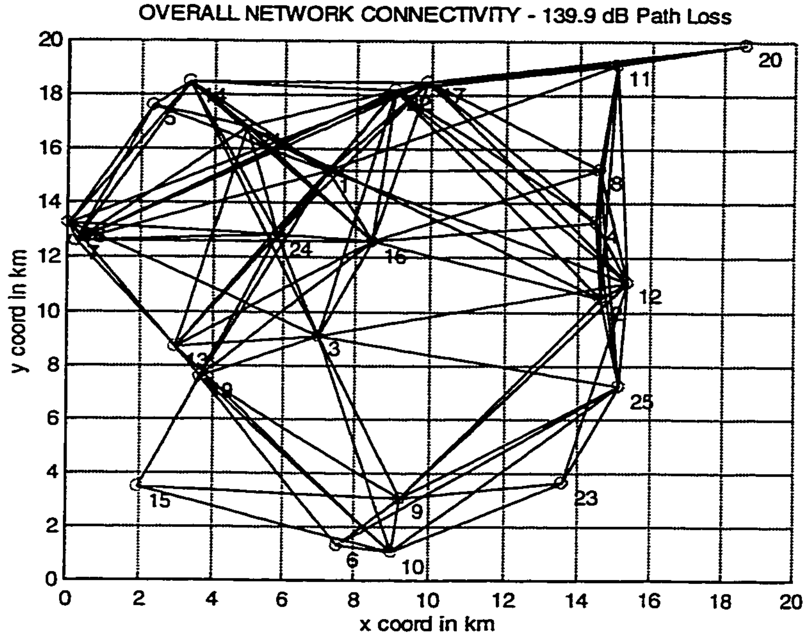
5.2b overall network connectivity
137.9 dB sustainable path loss
α data set



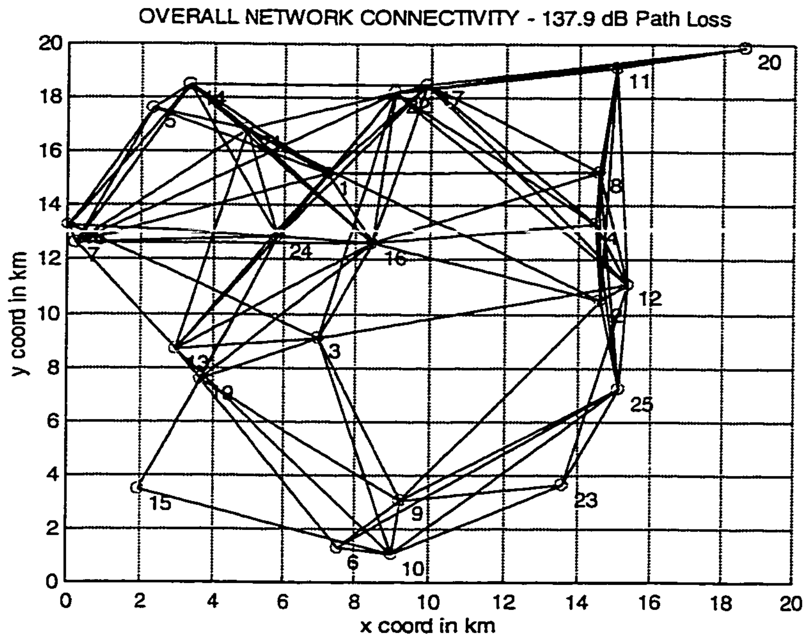
5.2c overall network connectivity
135.7 dB sustainable path loss
α data set



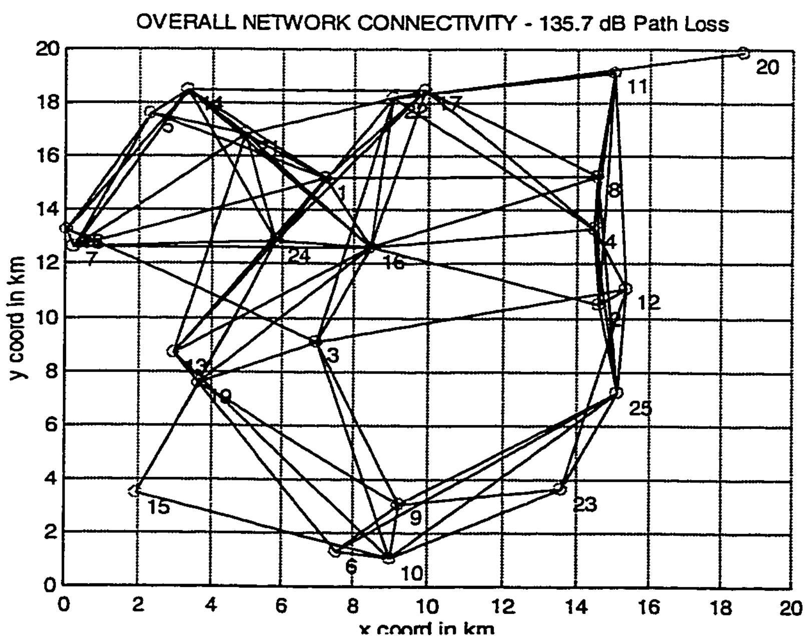
5.3a overall network connectivity
139.9 dB sustainable path loss
β data set



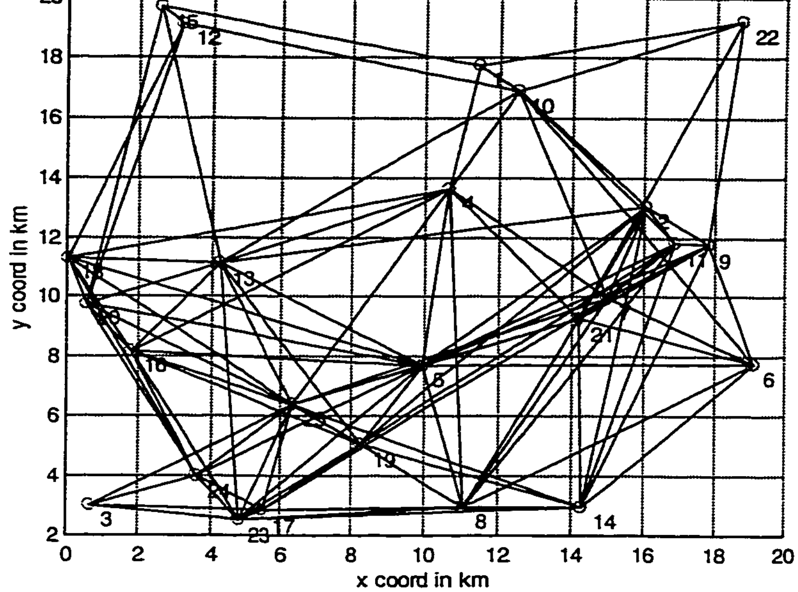
5.3b overall network connectivity
137.9 dB sustainable path loss
β data set



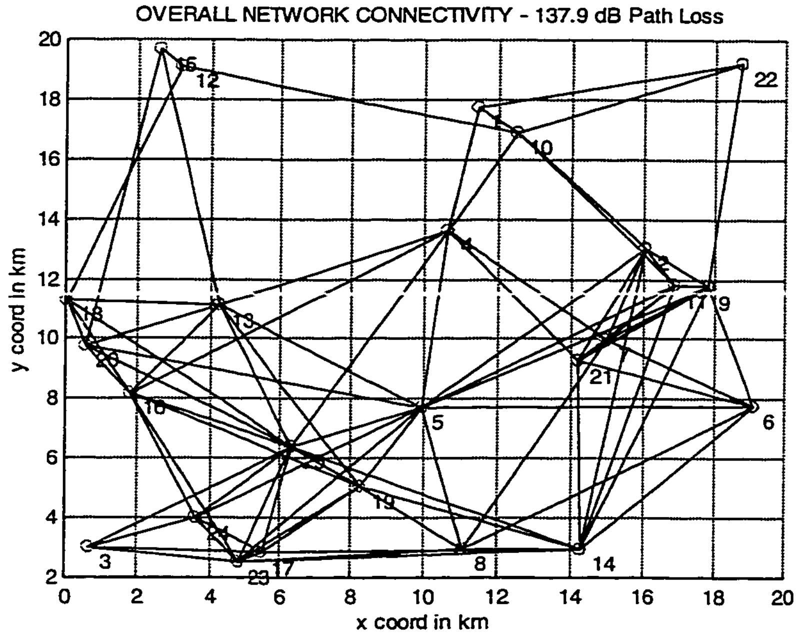
5.3c overall network connectivity
135.7 dB sustainable path loss
β data set



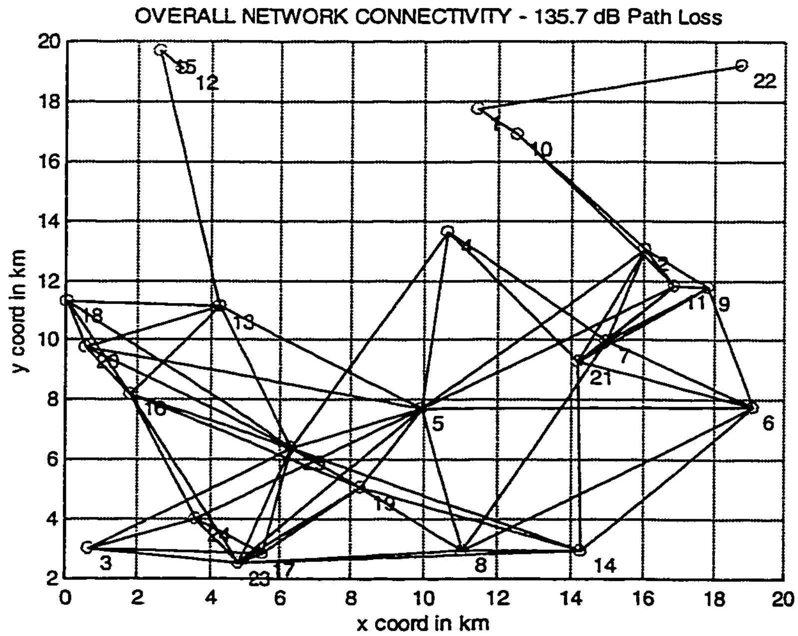
5.4a overall network connectivity
139.9 dB sustainable path loss
γ data set



5.4b overall network connectivity
137.9 dB sustainable path loss
γ data set



5.4c overall network connectivity
135.7 dB sustainable path loss
γ data set



5.1.7 Network Connectivity Observations

Figures 5.2, 5.3 and 5.4 serve to illustrate the differences in connectivity between the three options. Generally, in the case of simple-slotted CDMA there are many connections and many routes available to the network. The number of connections and routes drops in the mid-slotted examples and decreases by a larger margin in the max-slotted case.

Table 5.5 below summarizes the observations regarding connectivity taken from the five data sets using the three different cut-off or path loss thresholds. Note that while information from the δ and ε data sets are used below they were not included in the preceding diagrams.

	Number of links in network, average nodal degree					
Sustainable path loss	α data set	β data set	γ data set	δ data set	ε data set	Avg degree
139.9 dB Simple-slotted	104, 4.2	109, 4.4	108, 4.3	100, 4.0	88, 3.5	4.1
137.9 dB Mid-slotted	80, 3.2	94, 3.8	84, 3.4	82, 3.3	70, 2.8	3.2
135.7 dB Max-slotted	70, 2.8	82, 3.3	68, 2.7	67, 2.7	59, 2.3	2.7

Table 5.5: Number of links and average nodal degree in network for α-ε data sets and various slotted CDMA configurations

In Table 5.5, the number of links in each network is based on the total quantity of connections available for the given cut-off threshold, the nodal distribution, and log-normal shadowing. The average nodal degree refers to the average number of one-hop connections for nodes in each network. There is reasonable consistency in the data obtained for the number of links, although some fluctuations are observed. These fluctuations reveal the impact of node distribution and log-normal shadowing, particularly in the case of the ε data set.

In virtually all of the cases every node in the network is connected, although in some examples this could require a larger number of hops. The one exception is the max-slotted case for the ε data set. In that example 7 of the 25 stations are cut-off from the rest of the network, although they do have links with each other. Thus, we cannot generally guarantee overall connectivity for the max-slotted case.

The results viewed in the last column are more significant than they may appear at first glance. The average nodal degree of terminals in the simple-slotted case is 0.9 connections higher than the mid-slotted case, and 1.4 connections higher than the max-slotted case. In other words, the average node in a simple-slotted network will have at least one more one-hop neighbor than the next best case. As we wish to minimize the number of hops required for routing, such differences can become critical.

Detailed analysis of the suitability of each approach requires an in-depth investigation of how these configurations would support the system specifications. This analysis follows.

5.1.8 Integrated Multimedia Support Comparison - Simple, Mid and Max-Slotted CDMA

In order to provide a more objective comparison of the three options under examination, a random selection of two connection models (models A and B) will be applied to all three CDMA options and the α, β, and γ data sets. Each connection model will consist of eleven simultaneous connections, in the various data types as specified in Chapter 1. These eleven connections are referred to as S1, S2, S3, S4 (the four voice circuits), V1, V2, and V3 (three video circuits) and L1, L2, M1 and M2 (the data circuits)

This comparison will allow a view to be formed of how difficult it is to achieve the necessary connections under the different data rates and connectivity tables. While the higher data rate options offer more capacity at each terminal, they provide lesser connectivity. The comparison will allow a decision to be made as to whether or not the added capacity is really required.

These random selections are:

		Model A		**Model B**
- voice:	S1a	between node 15 and node 24	S1b	between node 3 and node 5
	S2a	11-25	S2b	20-11
	S3a	6-5	S3b	9-18
	S4a	19-7	S4b	1-13
- video:	V1a	9-12	V1b	22-21
	V2a	14-16	V2b	4-17
	V3a	13-20	V3b	23-12
- low speed data	L1a	8-10	L1b	10-15
	L2a	3-17	L2b	19-7
- medium speed data	M1a	23-18	M1b	6-24
	M2a	1- 22	M2b	2-14

Two of the many resulting examples are depicted in Figure 5.5. Figure 5.5a illustrates the provision of the speech channels for S1a, S2a, S3a and S4a for the α data set and the simple-slotted case while Figure 5.5b does so for S1b, S2b, S3b and S4b with the β data set. The active links are shown by a solid line, while possible (but unused links) appear as dotted lines.

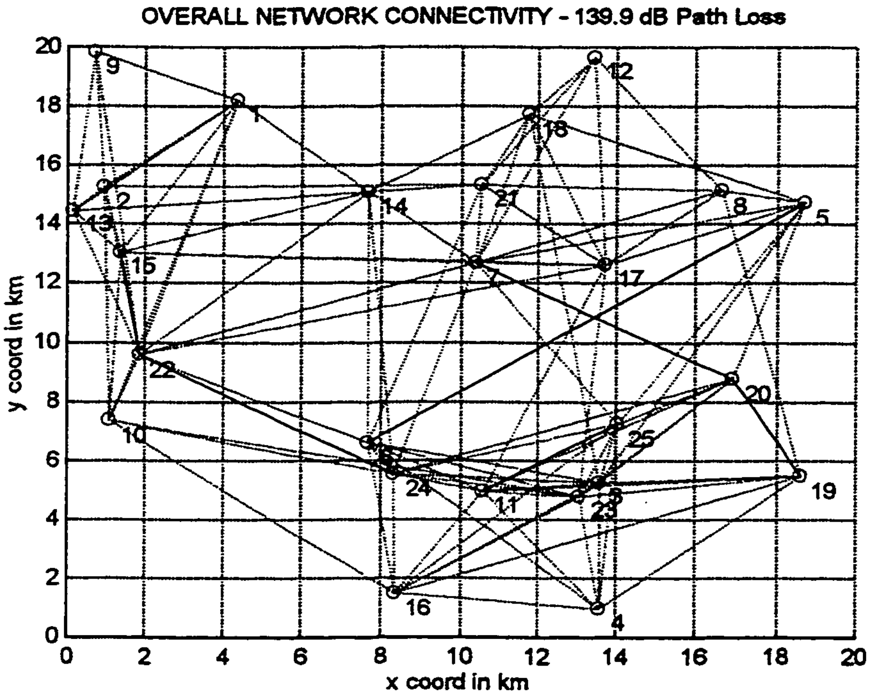


Figure 5.5 a: Voice routing for S1a, S2a, S3a, S4a, α data set, simple-slotted CDMA

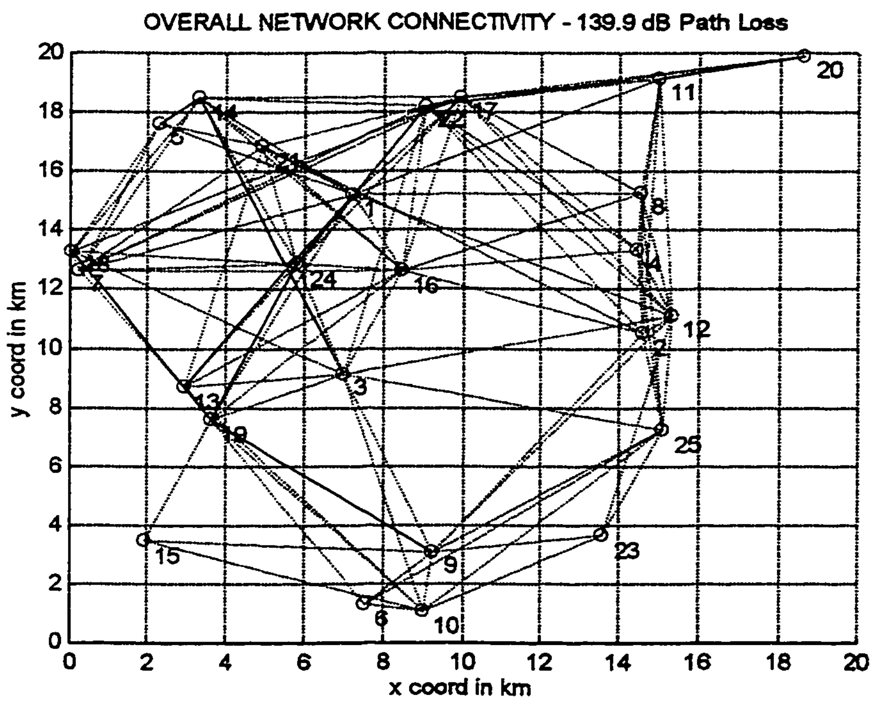


Figure 5.5b: Voice routing for S1b, S2b, S3b, S4b, β data set, simple-slotted CDMA

Predictably, the most interesting situations arising from these comparisons occur during the voice and video portions of the slotting. These are the busiest slots with the highest probability of conflict, because the network must support four and three simultaneous links, respectively.

It was observed in the connectivity maps that network coverage decreases as the data rate increases. One practical implication of this fact is that the average hop count from any given point to any other given point will rise. This is counterproductive, unless a shortage of connections makes the higher capacity systems necessary. One example of this is seen in the α data set for the simple and mid-slotted cases. For the simple-slotted case, nodes 3 and 5 are directly connected, but an intermediate node (25) is required for the mid-slotted case, thus increasing the number of hops required.

Figures 5.5a and 5.5b illustrate the possible routing to support the random connections in two of the examples. In fact, application of the random collection of connections listed above does not yield any situation that could not be supported under any of the three options. In other words, the simple-slotted CDMA example serves the users well for these examples.

Although application of the near-far problem makes it possible to devise situations where links could not be established due to high levels of interference, under routine operation this will not occur frequently. Thus, so long as terminal outages are discounted, given the connectivity diagrams in Figures 5.2a, 5.3a and 5.4a, it is difficult to devise situations where, for example, four simultaneous voice circuits could not ordinarily be supported in the simple-slotted CDMA scheme.

Thus, while the mid and max-slotted CDMA options presented here do provide additional capacity and generally feasible network connectivity, the added capacity is not needed (this assumes that no more than the specified circuit quantities were used). In addition, in view of the desirability of limiting the number of hops required to get from one hop to another, implementation of the higher capacity systems can be counter-productive.

Therefore, in view of the decreased connectivity of the higher rate cases and the ability of the simple-slotted case to handle the system's specified traffic, the simple-slotted CDMA infrastructure first identified in Chapter 4 and modified to carry control information will be selected for use in this network.

This conclusion means that we must have a data structure capable of supporting 183 cell/s through each node, with the cells interleaved in such a way as to ensure that delays incurred in the voice and video circuits are acceptable. For example, it would not be feasible to group all of the voice cells together, with 161 cells transmitted between the group of 22 voice cells as this would lead to unacceptable delays. The control cells, LSD, MSD, voice and video cells are distributed evenly throughout the control structure. This is illustrated below in a very simplified manner in Figure 5.6.

1	2	3	4	5	... ...	180	181	182	183

Figure 5.6: Network Data Structure, Simple-Slotted CDMA

Clearly, the cell structure is repeated every second. As long as the cells are interleaved in consistent and approximately even, the actual assignment of individual cells to specific data types is irrelevant. However, cell 1 could be node 1's first control cell, cell 2 could be a video cell (84 of the 183 cells will be video cells), node 3 a voice cell, node 4 a video cell, cell 5 node 2's first control cell and so on.

5.1.9 Peak Capacity

It should be noted that the stated system capacity for simple-slotted CDMA could be exceeded in several situations. For example, it would be possible in this network for up to 12 circuit pairs to conduct simultaneous voice conversations, as long as the resulting interference is at an acceptable level, and that no more than one conversation was supported by each terminal. The system has not been designed to support this situation and would not be in a position to provide service guarantees. Although its likelihood is small, if the conditions listed above were met the actual data capacity of the system would be even higher. This is another demonstration of the flexibility afforded by the CDMA architecture.

The simple-slotted CDMA structure provides nodes with the ability to simultaneously support several circuits. However, many issues remain for resolution to arrive at a more comprehensive network design. One of the critical questions that must be addressed concerns how paths are selected and packets are routed. Network routing will be addressed in the following section.

5.2 Routing for Distributed Packet Radio Networks

The design of an integrated multimedia network inherently introduces the overarching challenge of QoS provision. Virtual circuits required for voice and video channels should only be established if there is a reasonable likelihood that they can be maintained. For a wireless network, these challenges become much more difficult to address due to the band-limited environment the system must operate in, and the reduced control capabilities related to a wireless distributed architecture.

In order to satisfy the more stringent requirements of delay sensitive voice and video channels, resources must be reserved and controlled. The first step in establishing this resource reservation is the determination of the path or, more accurately, paths that the data can take. For connection oriented services, the application of an algorithm to select the path based upon a consistent, predetermined metric then allows resource reservation to occur. Resource reservation is, in turn, related to the concept of the control channel, which should be minimized in view of the system's bandwidth constraints.

This section will concentrate on considerations for routing in a distributed environment. A simple routing scheme for use in this network will be proposed, and a more complex algorithm will be explored for future use.

On a very basic level, routing methodologies can be placed in two broad categories: centralized and distributed approaches [16]. In centralized systems the routing decisions are made at central 'master' nodes or base-stations, whereas in distributed systems the decisions are made individually based upon information shared between nodes. Centralized approaches can be extremely effective in the computation of optimized routes, however they introduce significant disadvantages, namely poor survivability and responsiveness to network topology changes. Since this network must be survivable, only distributed routing will be considered.

5.2.1 Distributed Routing Approaches

Distributed methodologies offer superior survivability in the event of nodal outages. Furthermore, they are clearly more appropriate in the absence of centralized nodes providing 'control' functions for the overall network. However, the overhead required for protocols supporting the decentralized approach can be considerable.

Ideally, each node should be provided with the basic network connectivity information which is then processed by the routing algorithm to arrive at 'next hop' routing for each node in the system. Originating and intermediate nodes in the path must be able to determine which node it should forward a cell to in order for it to arrive at the intended destination terminal.

Basic decentralized or distributed routing methodologies can be categorized as flooding procedures or point-to-point protocols [1]. Flooding entails the transmission of cells to every node in the network, without any attempt to 'route' the cells efficiently. Flooding techniques do not generally require significant overhead and deal well with connectivity changes, but do tend to make poor use of system resources unless the intent actually was to broadcast the message to all nodes. Furthermore, for CDMA systems, the use of flooding must be reconciled with the system's code orientation, as will be seen in Chapter 6. Complex connectivity maps are not required in flooding networks, but care must be taken to prevent cells from endlessly re-transmitting or looping all over the network.

Point-to-point methods inherently involve a source and destination pair, with a route established to efficiently move the cells between the two. The control channel implications of point-to-point protocols are higher due to the requirement for each node to build a connectivity table to indicate the next hop required to arrive at a given terminal. However, this does allow for efficient routing of the data itself, and prevents information from disseminating to stations not requiring it.

Point-to-point transmission can take place in connectionless and connection oriented environments. Connection oriented environments are intended for applications that are delay sensitive, and the QoS applicable for VCs requires the dedicated reservation of bandwidth. Such is the case for voice and video connections where routes will be reserved along the selected path to ensure a guaranteed QoS.

For pure data transmission, connectionless approaches can be utilized, whereby individual cells do not necessarily have to travel the same route from the source node to the destination node. However it should be noted that the routing and scheduling of datagrams must include consideration of acknowledgements which may be facilitated by the grouping of cells (this will be discussed in Chapter 6). Connectionless schemes are ideal for networks with a rapidly changing topology. However they are not appropriate for constant bit rate applications such as voice and video.

For this design, a mix of connection oriented and connectionless circuits will be utilized for the establishment of the various multimedia services. Flooding will be reserved for those cases that warrant it, in other words for selected broadcast services (as detailed in Chapter 6).

5.2.2 Path Determination and Selection

Several point-to-point protocols are available to determine the optimum path for the connectionless and connection oriented circuits. The definition of 'optimum' can take many different forms, based upon the performance metric under evaluation. These will be examined in conjunction with considerations for routing in a distributed environment.

Any discussion of path selection makes it necessary to know what paths are available, a question intimately linked with the question of the path selection metric and the information available from the control channel. As mentioned, a distributed ad-hoc network is a freeform structure with no a-priori knowledge of the network's topology and therefore, the network must 'learn' its own structure on power-up. Further repetition during the control phase will allow for nodes to inform each other of their connectivity, the connectivity of their neighbors, and allows link information to be updated as terminals move and conditions change. This permits a gradual build-up and maintenance of mapping information.

The consolidated connectivity or mapping information can be interpreted based upon the empirical metric selected to judge links against each other. Several of these metrics will be identified shortly.

The initial convergence of the mapping information to the real world connectivity will take several cycles of the broadcast channel to achieve. The time required to do this is dependent upon the number of hops between extreme nodes in the network. This provides further justification for choosing a slotted CDMA scheme that minimizes the hop count in the network.

Due to the dynamic nature of the network, the delay associated with the promulgation of connectivity changes will also be dependent upon number of hops between extreme terminals in the network. Again, it will take some time for changes to cycle through the network. This requires the establishment of a path correction mechanism to handle cases where connectivity has changed, but a transmitting station is not yet aware of the change. This will be discussed in Section 5.4.

5.2.3 Path Delay

Before proceeding to examination of a simple routing protocol, it is reasonable to examine the path delay that would be incurred by the delay sensitive services. It is assumed that the various information carried by each terminal at 183 cells/sec is interleaved so that not all of the video or voice cells are grouped together (i.e. the 84 video cells that are transmitted every second are not sent consecutively, followed by 99 other cells).

In such a structure, one cell can be received every 5.46 milliseconds. If a video cell is received and is retransmitted on the next available video (out) cell, it will have to wait approximately 183/84 cells until it is retransmitted. This is because the 84 video cells are interleaved evenly among the 183 cells (at least in an approximate fashion, as illustrated in Figure 5.6). Thus, for a three hop half-duplex circuit, an end to end delay of approximately 36 ms will result, which is clearly within acceptable limits. For a half-duplex voice circuit with 22 of 183 cells (in and out), a three-hop connection will incur a delay of approximately 136 ms, again an acceptable number.

These delays are within acceptable limits, but they might not be if very long paths were selected. Thus, another argument appears in favor of minimizing the path length selected for VC services.

5.2.4 Routing Metrics

Once path options become known, it becomes necessary to select a path based on criteria applied consistently throughout the network. Such a network could be based upon empirical metrics such as link quality or the relative magnitude of the link margin, time delay, network traffic etc. Well known protocols exist to implement routing based upon such metrics, including the Bellman-Ford algorithm which will be examined shortly.

Some of the metrics are not easily applied to this network. For example, the delay encountered between different nodes should be very small in all cases as we are dealing with a small area as compared to the propagation speed. Traffic conditions are also not a critical consideration because the network has been designed to carry the traffic specified in Chapter 1 (peak situations could occur, but are not likely).

This network is small and relatively simple. Therefore, it is possible to implement a more basic routing protocol if simple criteria are also selected. The importance of hop minimization has been identified at several points. Shorter paths minimize the delay experienced by VC clients, and shorter paths maximize the connections available to other terminals attempting to communicate.

As will be seen, the selection of hop minimization as the basis for route selection will permit the establishment of a very simple routing protocol for use in this network. A simple structure to accomplish this will now be explored, followed by investigation of a more complex routing algorithm that could be used in future work.

5.3 Simple Routing Protocol Based on Hop Minimization

It is readily apparent that the importance of selecting the shortest paths available between nodes is paramount. This allows the implementation of a very simple routing protocol based on the construction of routing trees and the selection of minimum hop paths.

The information received by each terminal in the control channel was defined in Section 5.1. This information includes a binary list of one-hop neighbors and the known network connectivity from each point. It allows each node to build its own connectivity or routing tree which then allows decisions to be made for routing in view of the aim of doing so in a minimum number of hops.

Figure 5.7 provides a simple example of such a connectivity tree, for node 14 in accordance with the connectivity as shown in Figure 5.2a (α data set). The first level of nodes directly below node 14 is obtained by listening to its neighbor's 'pings', and is then transmitted in its own control channel. The second level of nodes is obtained from the corresponding one-hop lists transmitted by node 14's one-hop neighbors. Connectivity with nodes greater than three hops away (there are none in this example) would have to be established by determining which two hop neighbor has connectivity with it, from those node's network wide binary connectivity tables.

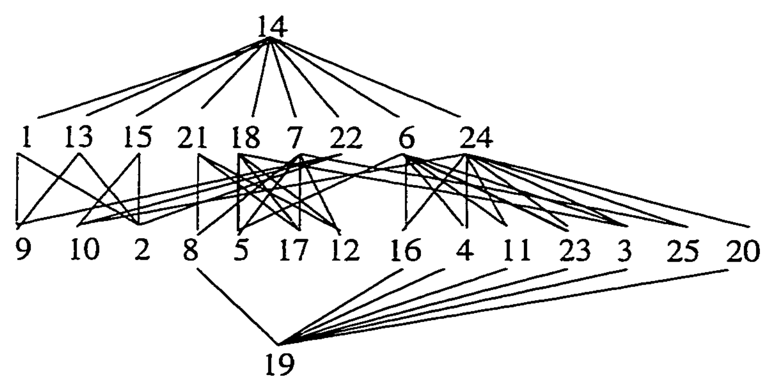


Figure 5.7: Sample connectivity/routing tree (data for node 14 from Figure 5.2a)

It should be noted that this diagram does not include some information that was omitted in the interests of clarity. Specifically, the addition of node-to-node links on the second and third level would be useful in obtaining other two and three hop links. For example, nodes 1 and 15 are connected, thus if the direct link from node 14 to 1 were to be interrupted, it would be possible to route a via two hops: 14-5 and 5-1. Node 1 would effectively drop to the third level of terminals from the top.

The sample tree shown above is easily constructed at each node based upon the connectivity received in the control channel. It is then simple to select paths based upon the shortest routes possible. It should be noted that this is really a selection of the shortest available paths, because for VCs there will have to be some form of reservation, and the shortest path that exists, or part thereof, may be reserved for use in part of other paths routes.

With the path selected, it is then possible to select the node to which the packet should be routed. This is either the destination node (for one-hop neighbors) or an intermediate node, which in turn will then have to go through the same process. It should be noted that for VCs, some form of reservation will be required to guarantee the necessary QoS. Therefore, once paths are determined they will only have to be changed when a change to the network's topology requires it.

In cases where connectivity changes, the affected nodes must attempt to reroute information based upon connectivity trees that have been rebuilt to integrate new information. This will be addressed in greater detail in Section 5.4

This simple hop minimization routing protocol is easily implemented so long as hop minimization remains the criteria by which routing decisions are based. The next section will briefly examine more complex algorithms that would allow other routing criteria to be considered.

5.4 Advanced Routing Algorithms Applied to Packet Radio

Given the premise that route computation should be primarily based upon hop minimization, it was possible to devise a simple path computation protocol through the construction of simple routing trees. However, any decision to implement a more complex path selection process would necessitate some other form of path selection protocol.

The motivation to use a more complex path selection method would be based on a desire to use more robust connections. In the simple protocol, there is no method used to judge the quality of one link versus another. Therefore, a two-hop link with each hop barely exceeding the cut-off threshold could be selected over a two-hop link with very large margins. The former would be far more likely to drop below the cut-off threshold if conditions were to adversely change (increasing the path loss) or if nodal movement were to result in increasing the path loss, resulting in a required path change.

Fortunately, algorithms exist to facilitate such routing decisions. These include the Bellman-Ford algorithm, Dijkstra's algorithm and the Floyd-Warshall algorithm. These algorithms are used in more complex routing protocols that are in wide use today, most notably on the Internet (e.g. Routing Information Protocol/RIP, GGP, EGP).

Some of these protocols do operate on the basis of distance minimization, albeit in a more complex manner, and are known as distance vector routing methods. Each method operates by different convergence methods. Of the three, the ubiquitous Bellman-Ford algorithm is the most common. This algorithm will be examined in greater detail.

Bersekas and Gallager provide a basic comparison of the Bellman-Ford algorithm, the Dijkstra algorithm, the Floyd-Warshall algorithm [16]. A specific example of the Bellman-Ford algorithm is provided here to demonstrate in general how the method converges.

5.4.1 The Bellman-Ford Algorithm

The Bellman-Ford algorithm is a widely used, basic method of finding the shortest paths of at most *h* arcs or hops from a source node to all other nodes [16]. The shortest path of hops $\leq h$ from a given node to node i is $D_i^{(h)}$. Initially $D_i^{(0)} = \infty$ for all $i \neq 1$ and $D_1^{(h)} = 0$ (i.e. the initial connectivity to other nodes is unknown, and the length to the path itself is zero).

In this algorithm, d_{ij} is the 'cost' of any point to point link between transmitting node i and receiving/target node j, and $d_{ij} = \infty$ for all unconnected nodes. The cost metric remains undefined but could apply, for example, to link quality with some quantities applied to each connection, indicating the magnitude of the existing signal to noise margin.

The algorithm runs as follows:

$D_i^{(h)} = 0$ (distance from all nodes to themselves is 0). For each successive $h \geq 0$:

$D_i^{(h+1)} = \min_j[D_j^{(h)} + d_{ji}]$ for all $i \neq 1$ (assuming node 1 used as the source)

Based upon the network portrayed in Figure 5.8 on the following page, and using node 1 as the source node, for one hop we have:

$D_1^{(1)} = 0$ $D_2^{(1)} = 1$ $D_3^{(1)} = \infty$ $D_4^{(1)} = 4$ $D_5^{(1)} = \infty$ $D_6^{(1)} = \infty$

For two hops:

$D_1^{(2)} = 0$ $D_2^{(2)} = 1$ $D_3^{(2)} = 4$ $D_4^{(2)} = 4$ $D_5^{(2)} = 2$ $D_6^{(2)} = \infty$

Note that $D_3^{(1)} = \infty$ but $D_3^{(2)} = 4$, indicating that there is no one hop path from node 1 to node 3, but there is a two hop path. To continue the process, we look at three hops:

$D_1^{(3)} = 0$ $D_2^{(3)} = 1$ $D_3^{(3)} = 3$ $D_4^{(3)} = 3$ $D_5^{(3)} = 2$ $D_6^{(3)} = 6$

Note that now $D_3^{(3)} = 3$, indicating that the three hop path from node 1 to node 3 effectively costs less than the two hop path, according to whatever cost metric was selected.

For four hops:

$D_1^{(4)} = 0$ $D_2^{(4)} = 1$ $D_3^{(4)} = 2$ $D_4^{(4)} = 9$ $D_5^{(4)} = 4$ $D_6^{(4)} = 5$

In order to determine the optimum paths from all source nodes to all other source nodes, the notation is then modified as follows. The optimum path from a source node i to a destination node j in a network with hop length h is $D_{ij}^{(h)}$. The algorithm is set as follows:

Initialize: $D_{ij}^{(0)} = \infty$ for all $i \neq j$ $\quad D_{ij}^{(h)} = 0$ for all $h \geq 0$

And then iterate: $D_{ij}^{(h+1)} = \min[d_{ik} + D_{kj}^{(h)}]$ for $i \neq j$, for all k.

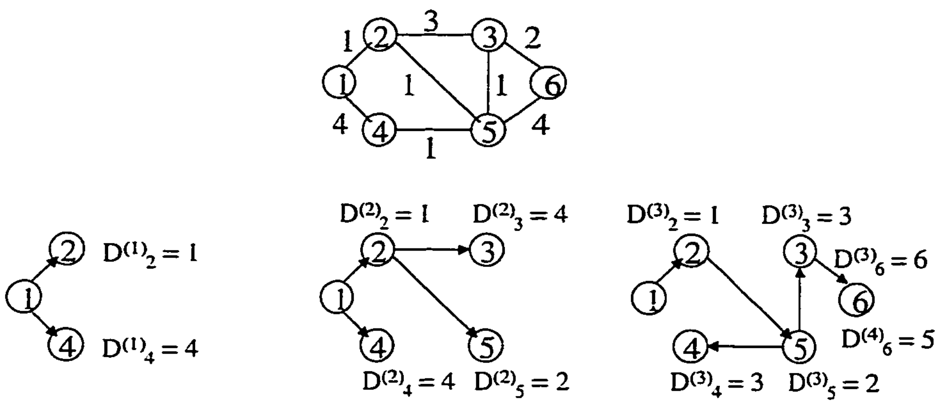


Figure 5.8: Bellman-Ford algorithm applied to a simple network [16]

It can be seen from the example that the Bellman-Ford algorithm simply computes the shortest path from a given node to all other nodes, first by looking at connections that are one hop in length, then comparing to connections two hops in length, three hops in length and so on. Part of the popularity of this algorithm, particularly for distributed applications, is that when arc lengths are positive (and they always will be for such a network), the iterations can be done in parallel for different nodes in virtually any order.

Dijkstra's algorithm, in contrast to the Bellman-Ford algorithm, finds the shortest path in order of increasing path length. Although this algorithm converges differently, it is still appropriate for distributed applications. This contrasts the Floyd-Warshall algorithm, which requires centralized control to be implemented and is therefore inappropriate for use in a distributed network.

Bersekas and Gallager indicate that the Bellman-Ford algorithm can be adopted in distributed networks without great difficulty [16]. However, to do so would require a firm definition of the cost metric used to determine d_{ij}. As suggested previously, a logical approach would be to assign a number to each link, inversely proportional to the magnitude of the link margin. This would allow paths to be selected that would minimize the requirement to set up new paths while connections are established and in use (i.e. the connections would be more robust). This is left as an area for future enhancement.

With examination of path determination and routing complete, the next step is to examine scheduling, reservation and the related issue of route maintenance in a changing topology, for both connectionless and connection oriented systems.

5.5 Scheduling, Reservation and Route Maintenance

In this network, scheduling and reservation must be carried out for two very different services: connection oriented, delay-sensitive voice and video traffic, and connectionless low speed or medium speed datagrams (neither are delay sensitive but both are error sensitive). The specific scheduling requirement for these two broad categories differ significantly.

The voice and video virtual circuits require that a predetermined QoS be provided and guaranteed, and that paths be reserved and maintained for the duration of the VC link. Such a scheme can be accomplished by utilizing the fast reservation scheme suggested by Gerla [13] and described below. These services contrast the datagram connections, which can be routed independently by groups of cells or individual cells (depending upon the acknowledgement scheme which will be discussed in Chapter 6). As datagram traffic is not generally time sensitive, this data can be buffered until transmit slots become available.

5.5.1 Fast Reservation for Virtual Circuits

Gerla's fast reservation scheme for voice and video traffic adds interesting elements to the mobile network architecture. It was proposed because more conventional virtual circuit reconfiguration methods fail. These protocols fail because they cannot keep pace with station movement and path outages [13].

The fast reservation scheme also largely eradicates the requirement for a Busy Tone Multiple Access architecture, because a VC is only opened if the necessary slots are reserved. Once the slots are reserved they will remain reserved for the VC until the VC is terminated. Therefore, there is no need to sense if the channel is busy, rather one must determine if unreserved slots are available at the source and destination nodes, at least in the case of voice and video traffic. This will require a slight modification to the network's control structure and will be pursued in Section 6.1.

In the fast reservation algorithm, VC cells are routed individually, but not independently. The first cell captures the required slots and reserves them at each terminal for the following packets, assuming that the slots are available. If slots on a given route are not available, then other paths must be investigated, and again slot availability must be confirmed. If no paths are available, then the VC cannot be initiated unless intermediate nodes closer to the source can initiate other new paths.

5.5.2 Connectivity Changes and VC Paths

Due to the dynamic nature of the network's topology, there is a clear requirement to periodically update connectivity information. Nodes are not static, so as they move, network connectivity will change and nodes may move outside of coverage. In addition, nodes can fail. Any of these conditions will result in a connectivity change which must be promulgated. If these connectivity changes affect active transmissions where paths have been reserved, then some corrective action to maintain connectivity must be taken.

In this network architecture, updates are provided every second, but with multiple hops between nodes on extreme edges of the network, it takes time for this information to cycle throughout. Thus, one must consider what happens if information is transmitted during a topology change, and what happens in the interim before a topology change is reflected throughout the network's routing tables (i.e. the routing table constructed at each node).

Fortunately, the architecture proposed here responds well to topology changes. In the case where a route is in the process of being established after a link has disappeared and the information has not yet been updated, then there is no significant complication. No path can be reserved, and the intermediate node can try alternative paths to the destination. If this does not succeed, the source node has the option of trying again by which time the routing table information will have aged if a slight delay is imposed.

In the event of an outage after a path has been established, several of the lead cells will likely be lost before the intermediate node becomes aware of the problem via the network's control channel. The next cell behind the outage can then be routed along a new path to the destination, reserving the required new slots. This assumes that a new path is available and that new slots are available for reservation. The new path may be routed through a different chain of intermediate nodes as required. In such a situation the loss of several voice or video cells is not catastrophic, although it will represent a nuisance for the user.

An example of an outage in an established VC is as follows: if the third intermediate node fails on a path, then the second intermediate node will try to find an available alternate path. If this node cannot resolve the problem, then the packets are returned to the originating node for resolution and so on. If there are absolutely no alternate paths, or if there are no available slots on the alternate paths, then the VC connection must be ceased.

Slots reserved on the receiver's side of an outage remain reserved for a fixed time interval. After the interval has passed, if no data has been passed on the reserved slots, they are released for use by other nodes. Thus the network can dynamically reconfigure its VCs and maintain promised QoS in all but the worst conditions. In this case the system has been designed to accommodate certain circuits in given quantities. Therefore it is should usually be possible to carry out VC reconfiguration 'on the fly' where necessary. This does not hold true if the network guarantees are exceeded (e.g. if more than three video VCs are initiated).

The actual mechanism that will be used to provide slot reservation for VC connections will be discussed in Chapter 6.

5.5.3 Scheduling of Datagrams

Datagram traffic introduces several different challenge related to scheduling: how are collisions minimized, how are route changes dealt with and how are acknowledgements (acks) implemented?

Unlike the VCs described above, data traffic in this network does not 'reserve' transmit space for QoS. Datagrams are transmitted individually as required and buffered where necessary (the assumption has been made that this traffic is not delay sensitive). Under normal circumstances data routing through a CDMA network can encounter collisions when two nodes attempt to send data to the same receiver. BTMA can be a solution where unreserved transmissions are taking place, but for CDMA this is problematic. BTMA use implies that the node wishing to send traffic must not only listen for its own traffic (i.e. its own PN code), it would also have to listen for the busy tone of the target node and would therefore require extra timeslots per node for BTMA transmission. Due to this added complexity, BTMA will not be used.

Fortunately, this system design largely obviates the requirement for datagram collision avoidance. First, the system is designed to allow for the transmission of one LSD and one MSD channel through each terminal. Therefore, within the system's data rate specifications and in view of the multitude of available routes, the only case in which there will usually be collisions is where two nodes will try to route traffic through the same node (or to the same node) at the same time. The implementation of an acknowledgement scheme (to be discussed in Chapter 6) will allow intermediate or originating nodes to recognize that there is a collision problem. In that case, the information is buffered and retransmitted after a slotted-Aloha delay to allow for retransmission to occur on a non-conflicting basis.

In the event that 'surge' or higher rate usage is required, buffers are used to hold information until slots become available. It may also be possible to pursue higher data rate options using free slots of other data types. This will be considered below and in Chapter 6.

A change in route availability for datagram traffic is less significant than for the VC case because no paths are reserved and the traffic is not time sensitive. Connectivity changes will result in rerouting in a manner virtually identical to the method outlined for VC traffic. Here however, cells must not be lost, again leading to the requirement for some form of acknowledgements.

In view of the possibility of collisions, and given the error sensitivity of the data traffic, there is a clear requirement to confirm the reliable receipt of any information that is sent. One possible method that will add very little overhead to the system is a passive acknowledgement scheme [13]. Using this methodology, the sending node listens for the onward transmission of its packet from its own target node to the next intermediate node or the destination itself. However, this simple scheme introduces a further complication. Since CDMA is in use, a specific code or PN sequence is assigned for each node.

The decision to use receiver oriented or transmitter oriented codes will have a significant impact on the feasibility to implement a passive ack scheme versus an active ack scheme. Acknowledgements, whether passive or active, will be received and this will determine the requirement for retransmissions (the only minor correction that may be required is regarding the requirement to 'listen' to another station's code in the case of a passive scheme). CDMA code orientation and its impact on broadcast messages and acknowledgements will also be discussed in Chapter 6.

While the selection of code orientation and the acknowledgement scheme remains to be resolved in the next chapter, it is appropriate to state that if collisions or errors are detected in datagram traffic, then the node must attempt to retransmit the data cell(s) again. Therefore, regardless of the acknowledgement scheme selected, the transmitting node is obligated to maintain a buffer of data transmitted in what could be termed an 'ack stack'. This will allow the recall of information for retransmission should no ack is received. If an ack is received, then the data is purged from the ack stack.

If two nodes simultaneously attempting to reach a third node cause a collision to take place, the automatic retransmission of the data payload will again result in collisions. Therefore, a variable back-off scheme is appropriate to help ensure that the data does not collide again. Since the data transmission scheme in this architecture is synchronized, it would be prudent to adopt a slotted, variable back-off retransmission scheme. In the event that repeated attempts fail, it is safe to assume that this is due to a nodal outage rather than collisions. Therefore, alternate routing should be attempted in a manner similar to that suggested for the VCs. First, the transmitting node will attempt to find another route that does not include the outage node. If this is unsuccessful, the datagram must be returned to the last intermediate node or the source node which will also attempt to find an alternate route.

Other issues related to the scheduling of datagram traffic also remain for resolution. Is there a way to provide a higher data rate if other system resources are not in use? Furthermore, is it possible to introduce a higher priority for urgent data traffic, rather than handling all data traffic on a first come-first served basis? It appears self-evident that these would be useful features. These potential capabilities will also be addressed in Chapter 6.

5.5.4 Basic Network Control Summary

Chapter 5 began with an analysis of the requirements for the control channel, resulting in a detailed specification of the control information to be transmitted amongst nodes to permit the construction of connectivity trees at each station. The network's data structure was further developed and a simple routing protocol was then defined, based entirely on the premise of route selection via hop minimization.

A brief exploration of the Bellman-Ford algorithm then followed to identify a method through which more comprehensive metrics could be applied. This would likely involve the application of a measurement inversely proportional to the signal to noise margin for each link. If this were done, established connections would be more robust, requiring fewer hand-offs due to exceeded cut-off thresholds.

Finally, a discussion of path reservation and scheduling took place for both connection oriented and connectionless services. This identified methods through which a path could be reserved, and a mechanism that could be used to reestablish routes that had been interrupted by outages or nodal movement. Figure 5.9 on the following page provides a summary of the activities that will occur in the receive chain (i.e. what does a node do with a received cell).

It should be noted that the network design is based on operation in normal circumstances, not worst case situations. The near-far problem can still occur if near interferers wash-out signals from more distant desired terminals. A very basic power control mechanism will be proposed in Chapter 6, however it will not eradicate the near-far problem. Therefore, it is acknowledged and accepted that there will be situations, albeit infrequent, in which the network may not operate as intended.

At this stage the network's multiple access scheme and architecture has been determined, the data structure established, the routing protocol selected, and basic scheduling and reservation issues resolved. Thus the next step is to integrate these features and address the ancillary issues raised in Chapters 4 and 5.

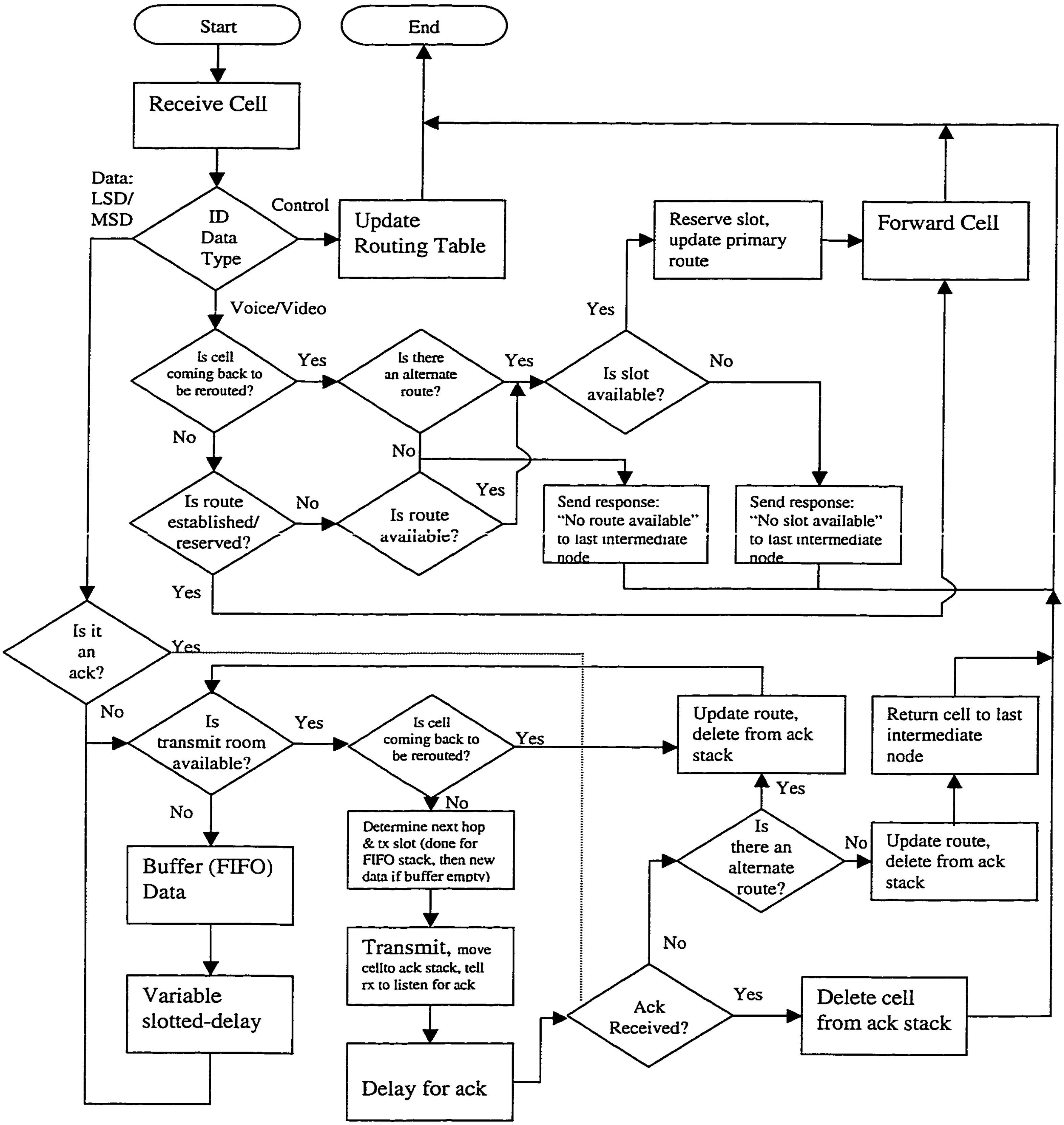


Figure 5.9: Receive Chain – Summary of Chapter 5

6.0 Network Integration & Enhancements

To this point, several of the critical areas in the network design have been determined by examining options and determining the optimum solution based upon the initial system specification, and the inherent capabilities of a distributed, wireless packet radio network. Specifically, a channel model has been developed, clustering architectures have been examined, multiple access techniques have been explored, and a network control and data cell structure selected. Routing and scheduling issues for this design have also been discussed.

These various elements must now be integrated to arrive at an overarching network architecture. A tentative architecture has already been developed. However, throughout the preceding chapters several issues related to the aforementioned have also been raised for resolution before the network design is finalized.

These issues include CDMA code orientation and its relationship with the control channel and acknowledgements, the requirement for flooding/broadcast, and the ability to provide services beyond the network's transport specifications (specifically higher rate data services on an available rate basis, and priority data broadcast). Each of these issues will be explored in turn, and an overall architecture will be proposed.

6.1 CDMA Code Orientation Related to Control Channel and Acknowledgements

With the decision to adopt the simple-slotted CDMA architecture comes the requirement to determine an 'orientation' for the PN codes as applied to the user terminals. This was briefly discussed in Sections 4.3, 4.4 and 5.5. This issue must now be resolved without detracting from previous decisions, with a view to also facilitating the provision of broadcast services if possible, in addition to acknowledgements.

One of the major problems experienced in large scale commercial CDMA networks is the provision of an adequate number of unique codes to permit network operation. With literally millions of potential CDMA cellular telephones, the simple application of one unique code per terminal is not practical because the PN codes simply do not exist in such quantities. However, this design involves the deployment of a relatively small number of terminals in a limited area and therefore the requirement for unique codes is orders of magnitude easier to satisfy.

As previously mentioned, Lin and Gerla recommended the use of transmitter oriented protocols, primarily because receiver oriented protocols cannot avoid intercluster collisions. This is true if a clustering algorithm is used, but in this case the only potential area of use is with the broadcast mode. and this has yet to be discussed. One major disadvantage of transmitter oriented protocols is that all nodes must listen for all potential transmitter codes rather than just listening for its own (and possibly a broadcast code) as is the case for a receiver oriented code. As the network will not always know who should be transmitting to a given node at a certain time, this is highly problematic.

6.1.1 Collision Risks – Receiver Oriented Protocol

With a receiver oriented protocol, there is the requirement for all nodes to know the PN code of all potential receivers, and in this design that quantity is 25. What are the collision risks for the architecture proposed thus far?

The simple-slotted CDMA architecture coupled with GPS synchronization provides discrete time slices for each node to transmit its 'ping' and connectivity table. Therefore, there is no possibility of a collision between control frames unless the network's synchronization is off, in which case the network will go down altogether. However, the actual transport of the control channel in a receiver oriented protocol must still be determined. This will be discussed below.

In the case of the connection oriented virtual circuits (voice and video), frame reservation takes place and no collisions will occur under routine operation provided that some mechanism has been added to ensure surrounding nodes 'know' that VC slots are reserved. The only exception to this is the case where collisions occur as two terminals attempt to reserve frames at a target node simultaneously. This will be extremely rare. When it does occur, nodes will have to retry because no connection will be established.

How do we ensure that surrounding nodes know of a given VC slot reservation? Busy Tone Multiple Access (BTMA) was previously mentioned as a possible solution, but its use introduces further complications with respect to code orientation. BTMA would have to either use a common flooding code or step through all individual receiver codes, and this would have to be applied individually for each set of VC slots. Alternatively, a separate channel could be added but this would add cost, complexity and could impact on LPI if CDMA were not used. What is suggested instead is an alteration to the control channel. This will be explored shortly.

The connectionless LSD and MSD datagrams may experience some collisions resulting in the loss of data. This was discussed in Chapter 5 and a mechanism has been determined to permit errors or losses to occur, and retransmissions to take place as required after a randomized slotted back-off delay if acknowledgements are not received. Datagram collisions are not likely, but when they occur, or when errors in transmission are encountered, the problem is rectified by the acknowledgement mechanism, whether it be active or passive. This will also be discussed below.

6.1.2 Control Channel Modification for Frame Reservation Awareness

The control channel specified in Section 5.1 includes several spare bytes. One of these bytes could readily be used to indicate, on a binary basis, which of its VC slots are currently reserved. Thus, when each node transmits its control frames, all one hop neighbors will know which VC slots are available. As this information is critical in scheduling and reserving routes, this mitigates almost entirely the requirement for a channel sensing mechanism such as BTMA.

Using this methodology, a station wishing to establish a path with another terminal will first check to see if the VC slot is free and will transmit only if this is the case. The receiving node must then change the state of its VC availability flag as it receives a cell during a VC slot.

One further complication is then raised. How is the control channel transmitted in a receiver oriented approach? The answer lies in the provision of a common network-wide broadcast code for the control channel. Thus, each node will have to listen for its own PN code and the common broadcast code for the control channel.

This solution is elegant because the control frames operate on fixed timeslots for each node, and therefore collisions will not occur. If a broadcast code is used and nodes listen for this common broadcast code in addition to its own unique receiver code, the necessary connectivity is achieved.

6.1.3 Acknowledgements

Several options are available to introduce an acknowledgement scheme to this architecture. Both passive and active ack mechanisms have unique advantages and disadvantages, all of which must be weighed in view of the requirements for this distributed wireless network

The introduction of the passive ack scheme for LSD and MSD data adds considerable simplicity to the broadcast chain. No added ack transmissions are required in a passive scheme, because after node A transmits to node B, A listens for the onward transmission from node B to node C. If it does not occur after a predefined period, then it is assumed that a retransmission is required.

Unfortunately, a passive ack scheme adds considerable obstacles and complexity to the receive chain in this design. This is due to the use of a receiver oriented code in the CDMA architecture. For multi-hop links, nodes that have transmitted datagrams will have to listen for the onward transmission of the datagram using the receiver code of the next intermediate node in the route. In addition to requiring the node to listen for additional codes, it would require that the node have knowledge of the next hop in the transmit chain beyond its own 'target' intermediate node. Otherwise, it would not know which additional code to listen for. As routing decisions for the datagrams are made in a fully distributed and independent fashion, this is highly problematic. Furthermore, in the case of one-hop links, there will be no onward transmission, and therefore no passive acknowledgement takes place.

These difficulties are very hard to overcome. One possible solution would be to have the intermediate node inform the originating node of the next hop once the routing decision is made. However, this immediately relegates this scheme from a passive one to an active one. Therefore, it appears fruitful to examine a true active scheme.

6.1.4 Active Acknowledgement Schemes

In an active scheme, the receiving node will transmit an acknowledgement stating that it has correctly received a datagram or series of datagrams. This is done on a link by link basis. In a reliable medium it is more efficient to transmit acks for a series of datagrams, thus reducing overhead. In this case, since we have used an RF channel model it should be appropriate to view the transmission medium as reliable. Given that one LSD and one MSD channel might be flowing through one station (albeit not on a VC basis), it should be possible to send the acks in response to each of these groups, once per second. The LSD and MSD channels flow at 6 and 21 cells per second respectively. If we added that capability to send two ack frames from each station per second, the requirement of the peak specification would be satisfied.

Accordingly, Table 5.1 is modified to read 7 and 22 cells respectively for the LSD and MSD, as opposed to 6 and 21. This will result in a negligible change to the system's connectivity diagrams as the overall transmit rate has increased from 183 to 185 cells per second.

It should be noted that with 48 byte payloads, these ack frames would be capable of identifying which data cells need to be retransmitted. Cell numbers will therefore be included in the header for all datagram cells. Although several other capabilities must be implemented via the header (e.g. error control coding, originating node, target node, intermediate nodes), this should be easily accomplished in view of the 7 byte header reserved for each frame or cell (55 byte frame with 48 byte payload). It should be noted that for this design the terms frame and cell are used interchangeably.

This modification requires that, as previously suggested, routing and acks for LSD and MSD packets must be done for groups of frames (specifically 6 LSD and 21 MSD frames) rather than on an individual frame by frame basis. To do otherwise would further increase the overhead requirement and this is not desirable.

To avoid the problem of continual retransmission loops in problem links, the addition of an ack counter is suggested. This would make it possible to indicate that a given retransmission is taking too long to succeed, and could allow for the return of the group of cells in question to the last intermediate link, for further transmission by a different route. This would also result in changes to Figure 5.8.

Due to the simplicity related to adding the critical ack capability in an active sense, link by link, in combination with a receiver oriented code, both methods will be used in this network design. Broadcast codes will remain in use for the control cells. Further alterations will be made as necessary for the other areas requiring rectification or improvement. This includes the ability to provide a broadcast capability for data, outside of the control channel, throughout the network.

6.2 Broadcast Capability Related to Code Orientation and Clustering

A system wide broadcast capability is highly desirable for every network. In a conventional network, broadcast is a highly efficient method of getting information quickly to every terminal. However, in a packet radio network where the topology is highly dynamic, such a capability is not as easy to implement.

It should be noted that the network design does already make use of a very limited broadcast capability in the control channel. However, this limited use of broadcast for control is markedly different from the requirements of a true data transport channel. Indeed, the control channels operate on discrete time slices for each terminal, and while each node uses a general broadcast code so that all other nodes in radio range can hear the transmitting terminal, there is no risk of collisions due to the time sliced configuration of the control mode.

In contrast, there are no such design defenses in this network for a basic implementation of a data broadcast mode. If we assume that a common broadcast code is used for the broadcast mode (as for the control channel), it is not difficult to forecast difficulties as illustrated in Figure 6.1 below.

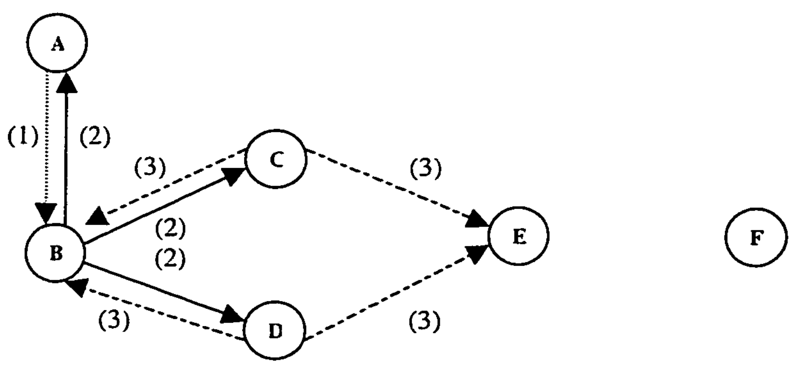


Figure 6.1: Simple Broadcast Conflict

Figure 6.1 demonstrates a very basic conflict that arises for simple broadcast as applied to a network using a common broadcast code. For connectivity, it is assumed that node A normally can contact node B, node B can normally contact nodes A, C and

D, nodes C and D can normally contact nodes B and E, node E can normally contact nodes C, D and F, and node F can normally contact node E. Note that this diagram does not include any depiction of the acknowledgements that would occur for datagram traffic.

Node A begins by transmitting a short broadcast message to node B (this is annotated as (1)). Node B then retransmits this message which is in turn received at nodes A, C and D (annotated (2)). A mechanism should be established for node A to recognize that it has just sent this message so that it can be disregarded. Nodes C and D then simultaneously retransmit the message using the same broadcast code (annotated (3)). This message should then be received at node E, but a collision will occur between the traffic sent from nodes C and D. No traffic can then be forwarded to node F.

It should also be noted that if node A continued to send broadcast traffic, there would be a conflict between frames transmitted from node A to B, and the re-transmissions from nodes C and D that are also received at node A. This is because a common broadcast code is used. In this case the functionality of CDMA technology is not used to advantage.

6.2.1 Priority Discrete Data Broadcast Mode

Herein lies a basic conflict experienced in CDMA networks. Receiver oriented codes are suited for point to point transmissions and this has been implemented here. Unfortunately, the application of receiver oriented codes to the broadcast domain introduces the high likelihood of collisions as the scheduling does not allow for rectification of the conflicts outlined in Figure 6.1.

Transmitter oriented codes are generally more suited for the broadcast domain because they allow terminals to listen for the PN codes of the common transmitter. In this design, all nodes cannot communicate with all other nodes, so customization is required. However, before proposing a solution, the problem needs to be scaled back to a more realistic goal. It would be ideal to provide voice or video broadcast, however, this requirement would significantly increase complexity and feasibility of any solution. Therefore, in this network, broadcast will be provided for simplex data transmission consisting of the sum total of the data cells, namely 7 + 22 = 29 cells per second in what we term a priority discrete data packet mode.

As no acks are included in this packet (they will be handled separately), 29 data cells (29 x 48 = 1392 bytes) will be available for broadcast in a discrete data packet. This could consist of a very short wave file, an overlay for a pre-stored map indicated in the preamble, or it would more likely be a text message.

Note that transmission of the discrete data frames occurs in one second, but is not analogous to 29 data cells per second because this discrete broadcast is permitted for only that one second prior to the resumption of normal data transmissions. This is done to eradicate collisions between new data flowing into a terminal and retransmissions from terminals farther down the line. Using this methodology and the previous example, node A will not have further transmissions, so retransmissions from nodes C and D will not collide with new transmissions from node A. With the one second limitation there are no new transmissions from node A.

The limitation of 29 data cells for any data broadcast eliminates one of the conflicts illustrated in Figure 6.1. However, one other major conflict remains, specifically the problem of nodes C and D transmitting simultaneously to node E. This complication can be addressed through the reintroduction of the concept of the clusterhead, and the selective use of transmitter oriented codes.

6.2.2 Clustering for Priority Discrete Data Broadcast

The addition of the priority discrete data broadcast mode to the network introduces a special case that must be handled in a unique fashion. The impending transmission of a priority discrete data broadcast packet is indicated by a flag set in the control frame. The originating node will set this flag in its control frame to state that it will send a priority broadcast packet (of 29 cells). This will be sent to that node's clusterhead, and will be followed by transmission of the data itself via use of the originating node's code. This is why this method is termed "Priority Discrete Data Broadcast". The discrete data packet will take priority over all other traffic for the short interval required for its transmission.

The clusterhead, after receiving the flag, will set its own flag which will indicate to subordinate nodes and other clusterheads (some of which are accessed through one hop gateways and multi-hop paths) that broadcast traffic is incoming. Those nodes will switch to a transmitter oriented code scheme for the data portion of the next timeslot to listen for that clusterhead's code. The clusterhead will then receive the broadcast packet of 29 cells, and transmit it. The transmission to other clusterheads will occur directly and by the proper paths to those clusterheads that are more than one hop away.

Adoption of this scheme would eliminate the collision of data moving from nodes C or D to node E. Node E would listen for the information to come from either its clusterhead or an intermediate node. Node F would receive the information in a similar manner. Thus CDMA is leveraged to prevent collisions even though multiple transmissions are occurring simultaneously.

Acknowledgements in this case will also occur on a link by link basis. Once a node receives the broadcast data, it will send an acknowledgement back to the node's clusterhead. Clusterheads will send acks back to the originating node. If any clusterhead does not receive an ack from one of its subordinate nodes, the clusterhead will make the appropriate retransmission. Similarly, the originating node will make the appropriate retransmissions to any clusterhead that does not make the proper ack.

After the discrete one second data broadcast is completed then the data frames are freed up (via another control flag) and buffered LSD/MSD traffic is allowed to resume, with the addition of a very limited requirement for datagram acks from the receiving nodes. However, the discrete data broadcast packet must remain buffered for some time to ensure that the packet can be retransmitted to nodes that do not ack receipt.

The discrete data broadcast proposed here is not simple, nor foolproof. It relies heavily upon the timeliness of topology information. The network design in general requires the ability for nodes to listen for multiple codes (either broadcast or the receiver's own code) and now for this methodology must also be able to listen for another node's code as the temporary switch is made to a transmitter oriented scheme.

This topology also requires clusterheads to be aware of the identity of all other clusterheads. This is not a problem if all nodes are connected to the network. It does however require another modification to the control channel.

In the control channel, each node periodically transmits information including its own ID, one-hop neighbors, information regarding other node's connectivity maps, and various flags which indicate VC reservation information and impending discrete data broadcast. There is now a requirement for clusterhead information to be transmitted. As we are already transmitting connectivity information, it is not difficult to efficiently integrate clusterhead information to the network connectivity matrix without adding significantly to the control channel size.

In addition, intermediate nodes between clusterheads must be aware that they are gateways between clusterheads. Assuming that some mechanism were implemented to allow this, the intermediate nodes between clusterheads would also have to listen for broadcast traffic.

Figure 6.2 illustrates the network hierarchy based upon clustering, as described in Chapter 3 with a 139.9 dB sustainable path loss and the α data set. It should be noted that proximity does not necessarily reveal cluster membership. This is demonstrated by node 18, which is associated with node 3 as its clusterhead although it is much closer to node 7.

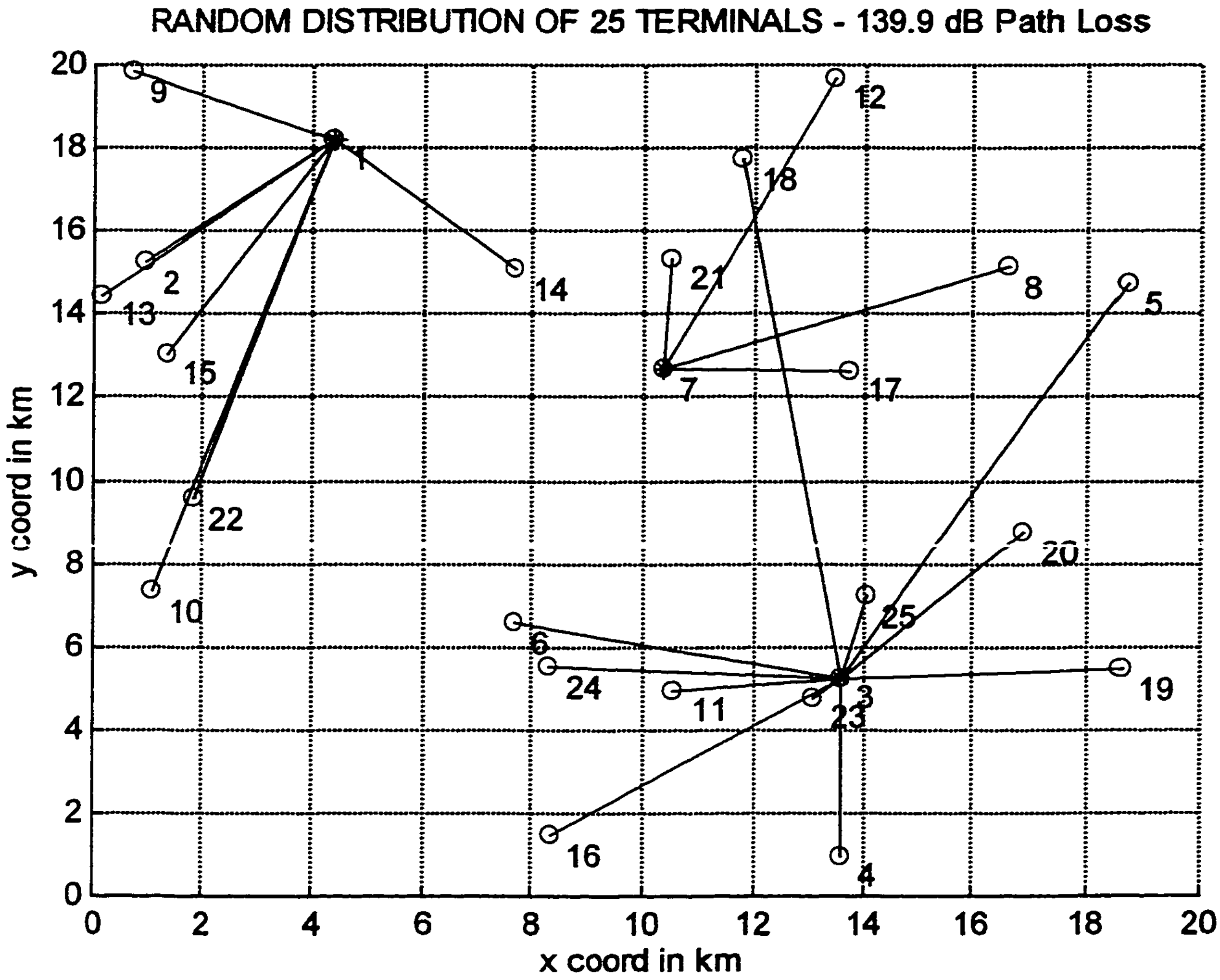


Figure 6.2: Network hierarchy for broadcast – 139.9 dB path loss, α data set

6.3 Other Capabilities Beyond Network Specification

Several capabilities that were not originally identified in the system specification have been identified earlier in this work and merit investigation. These capabilities include the possibility of establishing a voice only sub-network, higher rate data connections through the use of otherwise idle system capacity, power control and GPS enhancements.

6.3.1 Voice Sub-network

One of the intriguing observations that could be gleaned from earlier results is that for low rate pure CDMA (i.e. $R_b = 8$ kbps) the relatively high cut-off thresholds make it possible for network wide one-hop connectivity. This would permit the establishment of a voice sub-network if all other services were suspended.

Such action would permit the establishment of some form of emergency network if it were required. Although not pursued in detail here, it would not be overly difficult to implement so long as the transceiver was capable of operating at different rates. However, the existing connectivity tables for the higher rate system are quite good, thus such an enhancement is not critical. It would however permit network operation over a wider geographical range. Further implementation of this voice sub-network will be left for future research.

6.3.2 Available Rate Data

The network design has been based upon the ability to simultaneously carry different types of VC and datagram traffic. The data traffic is defined as low-speed and medium speed. However, it is apparent that under light traffic conditions, it should be possible to pass traffic at higher traffic rates by using other unencumbered portions of the system's data transport capacity, namely the space allotted to the various VCs.

This capability would be useful for moving larger quantities of data such as elaborate graphics or recorded full-motion video files. What would be the impact of making such a modification?

First, there would be the requirement to provide additional error control coding, or alternately more bandwidth or power. This is because, in accordance with Table 5.2, the BER for the VC traffic translates to an Eb/No of 6.8 dB as opposed to the more restrictive Eb/No for the datagram traffic, set at 9.6 dB. It is assumed that this change could be made with additional error control coding because this is relatively easy to accomplish with a CDMA system.

One simplification that can ease integration problems is that the available rate data will function only if no traffic is moving on the link in question. Thus, if no VC or datagram traffic is moving from node A to node B, then A could transmit at an available rate consisting of 185 cells/s –50 cells/s (control cells) = 135 cells/s or approximately 52 kbps. As each cell allows for seven bytes of overhead, there is a large amount of free bytes that could be used to provide added error control. In the interests of network consistency, this is viewed to be more desirable than making an additional increase to bandwidth or transmit power.

Using this methodology, the large groups of cells would move, hop by hop to the destination from the originator as links became available. It is clear that this method will not be appropriate for urgent traffic, or for use under heavy traffic conditions. It will also require additional data buffering at each terminal, and some form of acknowledgement mechanism with a longer delay window.

6.3.3 Power Control

One of the critical considerations in most CDMA designs is the power control mechanism used to mitigate the near-far problem. This design has been made at levels well below acceptable limits in terms of the number of simultaneous transmissions and as such it is not as critical. However, there is utility in limiting power where it is not required. This will decrease the impact of the near-far problem and increase battery life at the mobile terminals.

A specific example where simple power control can be useful is where the signals transmitted from the originator to the destination experience very little loss. This can be due to short distances or very permissive propagation conditions.

The system's control channels are used twice per second, and thus nodes will have a periodic indication of low loss paths. We assume that if node A's transmission to node B experiences very little loss, then the same can be concluded for the reverse path. In such a case, a reduction of 3 or 6 dBm in transmit power will conserve battery life, but will still ensure that necessary connectivity is maintained, while simultaneously decreasing the interference noise that other terminals will see as a result of this transmission.

Obviously this is not appropriate for any form of broadcast (e.g. control channels or priority discrete data), but it will apply for most point to point transmissions. Each node will make the decision to enact power control independently, based upon the signal strengths sensed for each one-hop node during the control channel. Thus, each terminal will have to store not only routing tables, but signal strength tables for all one-hop terminals.

This simple power control scheme does not eliminate the near-far problem because the transmitter does not take into account the impact on other receivers near-by, other than the intended receiver. A more complex power control mechanism would do this, but such a scheme is exceedingly difficult to accommodate in a fully distributed system, and therefore will not be considered.

6.3.4 GPS Information Broadcast

Several enhancement and additions have been made for the control channel. The existing size of the control channel (2 cells per node per second) is one limitation in the design. Some enhancements are easily implemented, whereas others such as the voice sub-network, would require significant additions.

One other system capability that could be of immense use in military applications would be the broadcast of terminal GPS information within the control channel. This would allow each terminal to form a real-time geographical layout of the other terminals in combination with preloaded cartographic files. However, this would require another significant change to the control channel. At a minimum, GPS information would be several bytes long per terminal. The transmission of the GPS information of 25 nodes cannot be accommodated in the present configuration and is left as another future enhancement.

The system is now integrated to the point of allowing full or limited functionality for a number of features. The key requirements for the various network elements are summarized in Section 6.4.

6.4 System Design Requirements Summary

The network architecture that has been designed in this and preceding chapters is dependent upon the provision of many key elements. This section will serve to summarize these key pieces in the network design.

6.4.1 Basic Mechanisms and Protocols

The network developed in general can be described as a 25 user distributed multimedia packet radio network:

- operating in the UHF band over a 20 km by 20 km piece of terrain;
- utilizing a slotted CDMA multiple access architecture with half-duplex virtual circuits incorporating voice and video traffic, connectionless low speed and medium speed datagram channels, and a network control channel integrated on a combined frame mechanism of 185 cells per second;
- incorporating a simple routing protocol at each node based on hop minimization applied to connectivity trees developed from control channel connectivity;
- establishing QoS for VCs via the use of a fast reservation scheme, and utilizing active acknowledgements to help ensure the integrity of datagrams. Both VC and datagram

services are scheduled via a quasi-bandwidth on demand approach (if slots are available for a VC then they are reserved, or in the case of a the datagram the frame is transmitted);

- permitting limited broadcast capability through the use of a priority discrete data broadcast mode, through the use of lowest ID clustering and the temporary suspension of other datagram transmissions; and,

- providing limited higher rate data services through the provision of available bit rate services (available when no other VC services are in use on a given link).

These capabilities require that the terminals be equipped with appropriate technology to permit the required functionality. Specifically, the terminals must be capable of:

- storing and updating various information including a network connectivity table, node clusterhead table, and PN code table (including the node's own code, all other user codes and the network broadcast code);

- storing transmitted datagrams (for low speed, medium speed and available rate transmission) and retransmitting that data if no acks are received (either to the same destination again, or after the ack counter has timed out, to either a new intermediate node or the previous intermediate node if no new path exists), or purging the data if acks are received;

- listening for multiple codes (broadcast code and unique terminal code), and transmitting multiple codes (transmitting using destination's code or transmitter's code (in the case of priority discrete data broadcast), and common code. Note this is not done simultaneously);

- running a simple routing protocol based on hop minimization;

- analyzing incoming signal strength (used to implement basic power control);

- running clustering algorithms based upon the lowest ID algorithm (for priority discrete broadcast mode);

- setting flags as appropriate in the control channel to indicate the availability of VC slots and the requirement to switch to priority discrete data broadcast mode;

- reserving available VC slots as required by incoming VC requests; and,

- transmitting at various power levels for basic power control when low-loss paths exist between transmitting and receiving stations.

7.0 Conclusion

The aim of this work was to develop a realistic network architecture capable of supporting a fully distributed wireless packet radio network. This is a highly complex task that is the subject of considerable effort by commercial telecommunications providers, academic institutions and governmental entities including the military. However, design assumptions made in Chapter 1 permitted the development of a high-level network design for specific applications.

7.1 Design Summary

Several key assumptions were made in Chapter 1 regarding the actual requirement itself. This was done to scope a manageable problem. As a result, several design decisions were also made to provide an initial starting point. These decisions were primarily dependent upon the application of this technology to the battlefield, but it is portable in the sense that other applications outside of the military are readily apparent. The requirements defined in Chapter 1 have been carried through the research when design options and considerations required resolution.

Chapter 2 focussed on the development of a realistic channel model that would provide a framework in which further analytical work and network development could be based. The channel model devised is based upon an implementation of the Hata empirical path loss model, and provides a non-linear simulation of the propagation conditions that can be experienced in an actual deployment. It should be noted that this differs from the vast majority of published work in the literature, because most other research was done based upon more simplistic, linear radio range approaches.

Chapter 3 concentrated on the linking and configuration of a packet radio network by investigating the primary linked-cluster approach that has been featured prominently in the literature, namely the lowest ID algorithm. Recent research indicates that, contrary to several earlier works, the introduction of clustering and the inherent clusterheads is counter-productive for most cases. However, the clustering concept was not entirely disregarded. In this design a limited clustering approach is used for one of the specific enhancements discussed in Chapter 6.

In Chapter 4 a basic analysis and comparison of TDMA and CDMA techniques was done with an emphasis on the requirements of the multimedia packet radio system specified in Chapter 1. Based upon the model used for comparison and the resulting connectivity thresholds and diagrams, slotted CDMA was selected for futher examination as the primary enabling technology. It was shown that slotted CDMA combined desirable spread spectrum properties with an element of TDMA time slicing. This permits scheduled and synchronized support to multiple circuits at the same terminal, while allowing simultaneous transmissions and maintaining interference rejection and LPI.

The bulk of the design work regarding the network architecture was carried out in Chapter 5. First, the network's basic data structure was investigated with a view to providing the specified voice, video and data services as specified in Chapter 1, in addition to the necessary control channel support. BPSK Eb/No requirements, based upon the bit error rates related to the various types of services, were then used to create link budgets for three slotted CDMA variants. This provided cut-off thresholds that could then be used in concert with the channel simulation model developed in Chapter 2.

The resulting connectivity diagrams helped to determine the overall sustainable path loss threshold that would be used. Simple-slotted CDMA with a WATM encapsulated data rate of 80.5 kbps was selected to permit the implementation of a control channel and simultaneous voice, video and data services throughout the network. More comprehensive CDMA and TDMA variants were shown to provide capacity that was simply not required based upon the connectivity available in the simple-slotted case.

The 80.5 kbps simple-slotted CDMA data structure was placed in a 183 cell per second framework. The emphasis in Chapter 5 then shifted to the routing of the various services. The network's control channel disseminates each node's connectivity information throughout the network. This control information is then used with a simple routing protocol based on the premise of hop minimization to determine the shortest routes available for intended destination nodes. A more complex routing solution, namely the Bellman-Ford algorithm was examined for use if more comprehensive path metrics were desired. This would prove useful for more robust connectivity if the routes were selected on the basis of a route metrics inversely proportional to link margin for each connection.

Scheduling and reservation was then discussed in Chapter 5, with a focus on the provision of these services for the two distinct information categories: virtual circuit (voice and video) traffic, and datagrams (low speed and medium speed data). Both types of traffic operate on a quasi-bandwidth on demand basis, with datagram transmissions occurring if free slots are available. In the case of VC traffic, fast reservation is utilized. Therefore, slot reservation along the route occurs prior to the establishment of voice or video circuits.

Chapter 6 presents issues required for network integration and enhancement. Many of the elements of this network are entirely dependent upon other design components for their implementation. For example, CDMA code orientation has a significant impact on routine data transport, the provision of acknowledgements and broadcasts. A receiver oriented design was selected for the normal operation of this network due to the simplicity and elegance of its integration with the network. Broadcast codes were also identified for use with the system's control channel. To mitigate collisions for VCs, a modification was made to the control channel to ensure one-hop neighbors are aware of terminal's VC slot reservation status.

Acknowledgements and their relationship with code orientation were also discussed in Chapter 6. An active ack scheme was selected for use with one second datagram groups (i.e. one ack for each 6 LSD cells or 21 MSD cells), thus resulting in an addition of two ack frames per second, thus resulting in a 185 frame/sec structure available at each node.

Chapter 6 continued with an investigation of a broadcast capability for the network. This presented a special case for resolution, and a unique 'priority discrete data broadcast' was proposed for broadcast purposes. This provides a limited capability to provide urgent short text messages or wave files of 29 cells throughout the network by temporarily suspending other datagram services and shifting to a transmitter oriented CDMA code. Other capability enhancements were also investigated. The feasibility of a voice sub-network for worst case system wide communications was hypothesized and explored, although it was not implemented here.

Available rate data services for higher speed data transport (at 135 cells/sec or 52 kbps effective link-by-link) was also proposed for use when VC traffic is very light. Power control was discussed for use when low loss paths are in use, and the ability of nodes to examine received signal power versus the cut-off threshold would allow the implementation of such a scheme. GPS information broadcast was also discussed as a potential enhancement although it was not implemented as, similar to the case of the proposed voice sub-network, it would have had a major impact on the system's control channel. Chapter 6 was completed with a summary of the system's requirements and the capabilities required at each network terminal.

7.2 Future Work

As mentioned above, many areas remain for future research endeavors in this field. This effort identified many areas that could lead to the improved operation of such a system:

- link costing and metrics. The implementation in this network of the Bellman-Ford algorithm would have permitted route optimization to be carried out on the basis of link quality or any other metric that was deemed desirable. With the simple routing protocol, each link, if it exists, is assumed to be equivalent. With the Bellman-Ford algorithm it would be possible to integrate some form of link quality metric to allow for enhanced link optimization, most likely a figure inversely proportional to the link margin associated with each connection. This could be coupled with statistical analysis in real time to permit the prediction of, for example, link outages, which would in turn allow for pre-emptive route changes to prevent the link outages;

- voice sub-network. The feasibility of a voice sub-network was identified above. Its presence would, for this design, prevent nodes from being entirely hidden, thus allowing for continuous simple connectivity. However, in view of existing connectivity the utility of a voice sub-network is limited unless one considers extremely harsh interference conditions or a geographically larger network operating range;

- GPS broadcast. The availability of GPS signals permitted the establishment of the highly synchronous system presented here. It would be valuable to further leverage this technology by allowing the nodes to transmit their real time location in the control channel. Again, this would require the addition of overhead to the control channel, but would allow for each node to create an physical accurate map of the other network users, which could be extremely useful in some applications including those beyond the military sphere, for example fleet management; and,

- gateway integration. This design has not integrated a gateway for connection of this network to the outside world, although clearly one would be desirable. In the network design all nodes are essentially 'equal' although some nodes are identified as clusterheads for use with the limited broadcast function. It is highly unlikely that such a network gateway would be based upon the node design presented here. For military applications, it is more likely that the network would be based around a mobile gateway station that would act as a router and connect to the outside world by satellite communications. It is not clear how this would impact issues such as routing within the network. Care would have to be taken to avoid negating the benefits of the CDMA technology such as low probability of intercept and resistance to narrow-band jamming. One possibility is the replacement of the one byte node ID with a six byte MAC address, but again this would require major growth in the control channel.

As demand for wireless multimedia services grows and commercial efforts in this field intensify, it is a virtual certainty that further great advances in wireless capabilities will occur. Indeed, many advances have been demonstrated in various test beds in recent years.

However, network designers will have to consider and weigh the impact of many of the limitations outlined here, such as the tradeoffs of mobility, efficiency, centralized routing, bandwidth and data transport capability. It will be fascinating to follow the advances in distributed, multimedia wireless designs, and their integration into what is increasingly an IP based, fiber driven world.

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Appendix 1

Matlab Script: Channel Model, Point Generator and Connectivity Determination

- 140 dB cut-off threshold, α data set example

```
hold on
freq = 900;
flag = 0;
adj = 0;
badj = 0;
cadj = 0;
rand('seed', 931316785);
randn('seed', 931316785);
points = 20 * rand(25,2);
distance = 0;
hatapathloss = 0;
adjhatapathloss = 0;
connectivity = 0;
linkcount = 0;
threshold = 140.2;
active = 0;
cluster = 0;
connections = 0;
number = 0;
c = 0;
d = 0;
e = 0;
f = 0;
x = 0;
y = 0;
hm = 1.5;
hb = 1.5;
ahm = 0;
k = 14.0;

a = log10(freq);
b = log10(hb);
pl1 = 69.55 +26.16*a - 13.82*b + ahm;
pl2 = 44.9 - 6.55*b;
pl3 = -4.78*a*a + 18.33*a - 40.94;
pl4 = pl1 + pl3 -k;
adj = 6.3 * randn(25);
```

```
for i = 1:25,
  number(i) = i;
  for j = i:25,
    if i == j
      distance(i,j) = 0;
      hatapathloss(i,j) = 0;
    else
      xd = points(i,1) - points(j,1);
      yd = points(i,2) - points(j,2);
      distance(i,j) = sqrt(xd * xd + yd * yd);
      hatapathloss(i,j) = pl4 + pl2*log10(distance(i,j));
      badj(i,j) = adj(i,j);
      if adj(i,j) < 0
        badj(i,j) = 0 - badj(i,j);
        flag = 1;
      end
      cadj(i,j) = 10 * log10(badj(i,j));
      if flag == 1
        cadj(i,j) = 0 - cadj(i,j);
        flag = 0;
      end
      adjhatapathloss(i,j) = hatapathloss(i,j) + cadj(i,j);
      if adjhatapathloss(i,j) < threshold,
        connectivity(i,j) = 1;
        connectivity(j,i) = 1;
      end
    end
  end
  x(i) = points(i,1);
  y(i) = points(i,2);
  cluster(i) = 1;
end
for k = 1:25,
  if cluster(k) == 1,
    plot(x(k),y(k),'y*');
    connections(k,k) = 1;
    connectivity(k,k) = 1;
    for l = (k+1):25,
      if cluster(l) == 1,
        if connectivity(k,l) == 1,
          cluster(l) = 0;
          connections(k,l) = 1;
          c(1) = x(k);
          c(2) = x(l);
          d(1) = y(k);
          d(2) = y(l);
          plot(c,d);
        end
      end
    end
  end
end
```

```
z = 1;
for i=1:25,
  if connectivity(i,i)==1,
    for j=(i+1):25,
      if connectivity(j,j)==1,
        for k=(i+1):25,
          if connectivity(i,k)==1,
            if connectivity(k,j)==1,
              gateway(z,1) = i;
              gateway(z,2) = j;
              gateway(z,3) = k;
              z = z + 1;
            end
          end
        end
      end
    end
  end
end

plot(x,y,'go')

for tx = 1:25,
  text(x(tx)+0.3,y(tx)-0.5,int2str(tx));
end

grid on
xlabel('x coord in km')
ylabel('y coord in km')
title('RANDOM DISTRIBUTION OF 25 TERMINALS - 140 dB Path Loss')
hold off
figure

hold on
for m = 1:25,
  for n = (m+1):25,
    if connectivity(m,n) == 1,
      e(1) = x(m);
      e(2) = x(n);
      f(1) = y(m);
      f(2) = y(n);
      plot(e,f);
      linkcount = linkcount + 1;
    end
  end
end
plot(x,y,'go')

for tx = 1:25,
  text(x(tx)+0.3,y(tx)-0.5,int2str(tx));
end
grid on
xlabel('x coord in km')
ylabel('y coord in km')
title('OVERALL NETWORK CONNECTIVITY - 150 dB Path Loss')
hold off
```

Appendix 2
Output - α Data Set, 140 dB Threshold

points = (x,y coordinates for nodes 1-25, top to bottom)			cluster = (if node is clusterhead then = '1')	
	4.3792	18.2064		1
	0.9409	15.2440		0
	13.5773	5.2491		1
	13.5859	0.9493		0
	18.6939	14.7216		0
	7.6700	6.5647		0
	10.3883	12.6528		1
	16.6193	15.1282		0
	0.6914	19.8207		0
	1.0692	7.3068		0
	10.5940	4.9408		0
	13.4230	19.6510		0
	0.1540	14.4532		0
	7.6683	15.0671		0
	1.3368	13.0304		0
	8.3497	1.4537		0
	13.7355	12.6327		0
	11.7795	17.6941		0
	18.6087	5.4542		0
	16.9233	8.7282		0
	10.5386	15.3299		0
	1.8393	9.5546		0
	13.0784	4.7555		0
	8.3200	5.4981		0
	14.0238	7.1853		0

connections = (Matrix indicates which nodes are clusterheads, and which nodes are connected to them. Any '1' in the leftmost column represents a clusterhead, and any '1' following to the right represent nodes which are subordinate to that clusterhead. For example, node 1 is a clusterhead, and node 2 is one of the nodes in node 1's cluster. For each column, a 1 will appear only once).

1	1	0	0	0	0	0	0	1	1	0	0	1	1	1	0	0	0	0	0	0	1	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	1	1	1	1	0	0	0	0	1	0	0	0	0	1	0	1	1	1	0	0	1	1	1
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	1	1	0	0	0	1	0	0	0	0	1	0	0	0	1	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0

Gateway = (indicates nodes which are one hop gateways between clusterheads. For example, node 2 is a gateway between nodes 1 and 13)

1	7	14
1	7	15
1	7	22
3	7	5
3	7	18
3	7	20
3	7	24
3	7	25

connectivity = (overall matrix that indicates which nodes are connected to each other. Row 1 is for node 1 and indicates connectivity with nodes 2, 9,10,15, and 22. This is based upon the threshold compared with the adjusted Hata path loss (adjhatapathloss). Note that any node that is not a clusterhead does not have a 1 in it own column/row (e.g. node 2 does not have a 1 in row 2 column 2). This is merely due to the programming convention.)

1	1	0	0	0	0	0	0	1	1	0	0	1	1	1	0	0	0	0	0	0	1	0	0	0
1	0	0	0	0	0	0	0	1	1	0	0	1	0	1	0	0	0	0	0	1	1	0	0	0
0	0	1	1	1	1	0	0	0	0	1	0	0	0	0	1	0	1	1	1	0	0	1	1	1
0	0	1	0	0	1	0	0	0	0	1	0	0	0	0	1	0	0	1	0	0	0	1	0	1
0	0	1	0	0	1	1	1	0	0	0	0	0	0	0	0	1	1	0	1	0	0	0	0	1
0	0	1	1	1	0	0	0	0	0	1	0	0	1	0	1	0	0	0	0	1	1	1	1	0
0	0	0	0	1	0	1	1	0	0	0	1	0	1	1	0	1	1	0	1	1	1	0	1	1
0	0	0	0	1	0	1	0	0	0	0	1	0	0	0	0	1	0	1	0	1	0	0	0	0
1	1	0	0	0	0	0	0	0	0	0	0	1	0	0	0	0	0	0	0	0	1	0	0	0
1	1	0	0	0	0	0	0	0	0	0	0	0	0	1	1	0	0	0	0	0	1	1	1	0
0	0	1	1	0	1	0	0	0	0	0	0	0	0	0	1	1	0	1	1	0	0	1	1	1
0	0	0	0	0	0	1	1	0	0	0	0	0	0	0	0	1	1	0	0	1	0	0	0	0
1	1	0	0	0	0	0	0	1	0	0	0	0	1	1	0	0	0	0	0	0	1	0	0	0
1	0	0	0	0	1	1	0	0	0	0	0	1	0	1	0	0	1	0	0	1	1	0	1	0
1	1	0	0	0	0	1	0	0	1	0	0	1	1	0	0	1	0	0	0	0	1	0	0	0
0	0	1	1	0	1	0	0	0	1	1	0	0	0	0	0	0	0	1	0	0	0	1	1	1
0	0	0	0	1	0	1	1	0	0	1	1	0	0	1	0	0	1	0	0	1	1	1	0	0
0	0	1	0	1	0	1	0	0	0	0	1	0	1	0	0	1	0	0	0	1	0	0	0	0
0	0	1	1	0	0	0	1	0	0	0	0	0	0	0	1	0	0	0	1	0	0	1	0	0
0	0	1	0	1	0	1	0	0	0	1	0	0	0	0	0	0	0	1	0	0	0	1	1	1
0	1	0	0	0	1	1	1	0	0	0	0	1	1	1	0	1	0	0	0	0	0	0	1	0
1	1	0	0	0	1	1	0	1	1	0	0	1	1	1	0	1	0	0	0	0	0	0	1	0
0	0	1	1	0	1	0	0	0	1	1	0	0	0	0	1	1	0	1	1	0	0	0	1	1
0	0	1	0	0	1	1	0	0	1	1	0	0	1	0	1	0	0	0	1	0	1	1	0	1
0	0	1	1	1	0	1	0	0	1	1	0	0	0	0	1	0	0	0	1	0	0	1	1	0

distance = (matrix denoting distance in km from one point to another. Take node numbers, use higher number as column, and lower number as row. Thus the distance from node 5 to node 2 is found in column 5, row 2 = 17.76 km. This calculation is later used in the Hata path loss calculation)

0	4.54	15.89	19.56	14.73	12.10	8.18	12.62	4.03	11.39	14.12	13.21	1.12	6.73	2.25
0	0	16.11	19.08	17.76	10.98	9.80	15.68	4.58	7.94	3.00	14.40	16.28	11.46	14.50
0	0	0	4.30	10.77	6.05	8.06	10.34	19.45	12.68	4.99	18.70	19.05	15.31	17.20
0	0	0	0	14.69	8.16	12.13	14.50	22.86	14.04	12.70	7.22	18.54	11.03	17.44
0	0	0	0	0	13.71	8.56	2.11	18.71	19.12	3.34	14.30	10.90	8.50	9.05
0	0	0	0	0	0	6.67	12.39	14.98	6.64	7.71	7.63	10.39	3.64	9.06
0	0	0	0	0	0	0	6.70	12.06	10.74	11.84	5.54	16.48	8.95	15.43
0	0	0	0	0	0	0	0	16.60	17.41	17.87	12.73	5.39	8.44	6.82
0	0	0	0	0	0	0	0	0	12.52	9.81	17.46	7.20	10.19	5.73
0	0	0	0	0	0	0	0	0	0	0	14.98	14.12	10.54	12.29
0	0	0	0	0	0	0	0	0	0	0	0	14.25	7.36	13.78
0	0	0	0	0	0	0	0	0	0	0	0	0	7.54	1.85
0	0	0	0	0	0	0	0	0	0	0	0	0	0	6.65
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0

17.22	10.89	7.42	19.11	15.72	6.80	9.02	16.02	13.31	14.65
15.65	13.06	11.11	20.20	17.26	9.60	5.76	16.04	12.22	15.37
6.46	7.39	12.57	5.04	4.83	10.53	12.50	0.70	5.26	1.99
5.26	11.68	16.84	6.75	8.46	14.70	14.56	3.84	6.96	6.25
16.82	5.38	7.53	9.27	6.25	8.18	17.63	11.44	13.88	8.87
5.16	8.58	11.86	10.99	9.50	9.22	6.55	5.70	1.25	6.38
11.38	3.35	5.23	10.93	7.62	2.68	9.09	8.34	7.45	6.57
15.98	3.81	5.48	9.88	6.41	6.08	15.80	10.96	12.71	8.36
19.90	14.89	11.29	22.97	19.66	10.82	10.33	19.50	16.23	18.37
9.34	13.74	14.92	17.64	15.92	12.41	2.38	12.28	7.47	12.96
4.15	8.31	12.81	8.03	7.38	10.39	9.90	2.49	2.34	4.10
18.89	7.03	2.56	15.11	11.47	5.20	15.37	14.90	15.04	12.48
15.37	13.70	12.07	20.53	17.72	10.42	5.18	16.16	12.12	15.66
13.63	6.54	4.88	14.56	11.22	2.88	8.02	11.64	9.59	10.12
13.54	12.40	11.44	18.86	16.17	9.48	3.51	14.36	10.27	13.97
0	12.41	16.60	11.01	11.24	14.05	10.39	5.77	4.04	8.07
0	0	5.43	8.68	5.04	4.18	12.29	7.90	8.96	5.46
0	0	0	14.02	10.34	2.67	12.85	13.00	12.68	10.75
0	0	0	0	3.68	12.75	17.26	5.57	10.29	4.90
0	0	0	0	0	9.18	15.11	5.53	9.19	3.28
0	0	0	0	0	0	10.44	10.88	10.08	8.86
0	0	0	0	0	0	0	12.22	7.65	12.41
0	0	0	0	0	0	0	0	4.82	2.61
0	0	0	0	0	0	0	0	0	5.95
0	0	0	0	0	0	0	0	0	0

Hatapathloss = (denotes Hata path loss from one node to another as calculated using the distance matrix above, and is read as for the distance matrix)

0	130.63	154.44	158.39	153.00	149.26	141.83	150.06	128.35	148.11	152.89	143.97	134.80	130.67	135.95
0	0	154.70	157.92	156.55	147.42	145.25	154.18	130.82	141.25	152.19	150.97	103.97	138.12	117.29
0	0	0	129.60	147.04	136.10	141.55	146.27	158.28	150.14	122.76	152.57	154.89	148.23	152.70
0	0	0	0	152.95	141.77	149.31	152.70	161.34	152.08	132.43	157.53	157.88	153.73	155.95
0	0	0	0	0	151.64	142.68	116.12	157.54	157.95	150.18	139.44	157.37	147.50	156.21
0	0	0	0	0	0	137.94	149.71	153.32	137.87	124.83	152.43	147.27	142.56	143.74
0	0	0	0	0	0	0	138.04	149.20	147.00	140.71	140.50	146.37	126.42	143.76
0	0	0	0	0	0	0	0	155.27	156.17	148.84	134.41	155.13	143.53	153.87
0	0	0	0	0	0	0	0	0	149.91	156.67	150.23	133.91	142.42	138.37
0	0	0	0	0	0	0	0	0	0	145.28	156.23	139.41	145.99	135.06
0	0	0	0	0	0	0	0	0	0	0	153.32	152.20	146.64	149.56
0	0	0	0	0	0	0	0	0	0	0	0	152.37	139.81	151.73
0	0	0	0	0	0	0	0	0	0	0	0	0	140.27	113.58
0	0	0	0	0	0	0	0	0	0	0	0	0	0	137.89
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0

155.96	147.26	139.97	157.94	154.24	138.31	143.67	154.59	151.07	152.89
154.15	140.71	147.64	159.00	156.01	144.86	135.16	154.62	149.46	153.80
147.34	139.88	149.99	132.61	131.80	146.62	149.88	95.17	133.45	114.94
133.44	148.60	155.54	138.16	142.47	152.96	152.78	127.45	138.75	136.71
155.52	133.86	140.24	144.19	136.71	141.82	156.41	148.19	151.87	143.35
133.05	142.73	148.89	147.44	144.67	144.10	137.61	134.97	106.12	137.11
148.10	124.85	133.32	147.32	140.48	120.63	143.83	142.20	140.04	137.65
154.55	127.32	134.20	145.40	137.18	136.20	154.33	147.38	150.20	142.23
158.71	153.21	147.95	161.44	158.48	147.14	146.26	158.33	154.84	157.19
144.35	151.68	153.24	156.42	154.47	149.74	118.34	149.54	140.11	150.56
128.92	142.12	150.34	141.47	139.86	146.37	145.44	119.24	118.06	128.70
157.73	138.93	119.72	153.49	148.25	133.20	153.80	153.22	153.40	149.85
153.80	151.62	149.21	159.31	156.51	146.42	133.14	154.76	149.29	154.16
151.52	137.56	132.00	152.78	147.82	122.00	141.45	148.53	144.85	145.88
151.39	149.73	148.19	157.69	154.77	144.63	125.76	152.52	144.15	151.99
0	149.74	155.27	147.47	147.87	152.10	146.37	135.18	128.44	141.55
0	0	134.02	142.94	132.62	129.08	149.55	141.17	142.55	134.13
0	0	0	152.05	146.27	120.55	150.40	150.63	150.15	147.01
0	0	0	0	126.66	150.26	156.01	134.54	146.18	132.09
0	0	0	0	0	144.02	153.48	134.38	144.03	124.49
0	0	0	0	0	0	146.46	147.23	145.79	143.34
0	0	0	0	0	0	0	149.45	130.54	149.75
0	0	0	0	0	0	0	0	131.76	120.10
0	0	0	0	0	0	0	0	0	135.77
0	0	0	0	0	0	0	0	0	0

Adjhatapathloss = (adjusted Hata path loss from above, as modified by Gaussian random variable, with standard deviation $\sigma = 8$ dB)

0	135.73	146.04	166.04	153.95	151.91	147.03	144.23	121.89	136.10	156.87	149.25	139.01	136.84	133.67
0	0	163.20	147.34	161.98	159.51	144.60	145.78	134.33	138.30	144.98	153.71	105.90	143.02	108.95
0	0	0	123.03	138.65	131.50	142.52	156.21	164.42	144.42	126.13	155.10	161.36	155.26	163.74
0	0	0	0	146.31	131.70	159.32	144.50	162.68	155.29	125.84	165.35	163.88	144.89	166.83
0	0	0	0	0	139.38	135.75	120.37	155.33	163.42	159.20	150.63	149.41	151.11	146.39
0	0	0	0	0	0	145.42	142.45	163.87	145.40	131.89	162.25	157.73	138.57	154.61
0	0	0	0	0	0	0	135.11	143.12	145.98	147.61	139.60	142.55	133.78	136.25
0	0	0	0	0	0	0	0	153.66	145.98	158.22	124.78	147.54	143.05	151.05
0	0	0	0	0	0	0	0	0	159.52	161.08	152.14	125.94	153.14	143.81
0	0	0	0	0	0	0	0	0	0	154.53	148.92	148.18	146.35	131.96
0	0	0	0	0	0	0	0	0	0	0	161.95	152.18	157.68	168.82
0	0	0	0	0	0	0	0	0	0	0	0	164.68	146.33	160.60
0	0	0	0	0	0	0	0	0	0	0	0	0	130.52	103.56
0	0	0	0	0	0	0	0	0	0	0	0	0	0	129.61
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0

144.93	147.01	146.03	152.84	159.47	142.53	134.38	148.25	147.34	163.09
159.06	156.04	155.29	169.77	150.13	138.16	132.83	161.71	150.10	151.72
127.43	141.93	141.08	130.58	133.60	152.19	159.31	94.53	137.97	107.88
133.96	145.73	161.76	137.75	141.44	153.39	152.63	131.51	145.03	139.83
157.41	129.67	134.40	152.10	135.22	141.87	163.44	157.29	163.94	133.33
130.79	151.28	154.40	156.54	140.76	138.24	138.67	139.50	114.82	145.78
142.55	128.76	127.47	142.08	131.08	118.64	139.59	143.43	130.91	131.46
153.83	137.37	143.10	139.00	145.49	127.74	163.18	150.14	146.08	146.08
167.53	160.37	155.91	154.10	167.51	154.95	135.83	164.41	144.18	156.47
138.30	161.34	146.20	161.65	159.34	144.13	131.02	139.05	132.57	140.53
134.73	132.73	144.63	139.90	132.06	141.33	146.13	118.20	126.74	120.88
151.78	136.79	129.99	157.24	152.79	125.28	160.04	144.30	163.47	148.99
159.37	159.60	157.00	168.17	147.53	142.02	128.82	160.36	160.94	161.66
144.93	148.19	121.16	161.18	149.91	132.15	132.87	156.23	137.74	149.29
160.15	139.40	153.28	150.39	159.01	149.87	115.04	161.05	153.45	158.79
0	147.12	159.21	139.91	156.04	140.84	155.85	127.62	122.86	133.94
0	0	126.46	149.38	141.44	121.79	139.87	136.06	152.89	143.85
0	0	0	156.88	153.19	116.28	143.33	159.35	158.74	158.32
0	0	0	0	117.95	142.14	155.08	135.55	142.40	140.61
0	0	0	0	0	130.60	144.96	130.77	132.27	114.93
0	0	0	0	0	0	147.01	146.71	148.68	147.65
0	0	0	0	0	0	0	149.72	131.67	153.27
0	0	0	0	0	0	0	0	123.41	115.00
0	0	0	0	0	0	0	0	0	138.40
0	0	0	0	0	0	0	0	0	0