The University of Kansas

Information and Telecommunication Technology Center

Technical Report

Rapidly Deployable Radio Network (RDRN) - Phase II

Final Report

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ITTC-FY2003-13380-15

September 2002

Abstract

The primary objective of the RDRN project was to create architectures, protocols, and prototype hardware and software for a high speed network that can be deployed rapidly in areas of military conflicts or civilian disasters where communication infrastructures are lacking and or destroyed (e.g. Desert Storm, Bosnia, Hurricane Andrews, LA, Turkey, Taiwan earth quake).

The rapid deployment requirement coupled with higher speed requirements and seamless integration with other commercial networks has lead to an approach that uses wireless technology for the communication links and ATM for networking. The Rapidly Deployable Radio Network (RDRN) being developed by the University of Kansas is a wireless ATM network and it consists of portable (mobile) communication nodes which can be deployed on the ground or on mobile platforms such as trucks, helicopters or fixed wing aircrafts. When deployed, the nodes use GPS derived location information to automatically configure themselves into a high capacity, fault tolerant network infrastructure.

Table of Contents

1.0 Introduction 4
1.1 Objective 4
1.2 Approach 4
2.0 Adaptive Communication Techniques 6
2.1 Channel Estimation 6
2.2 Adaptive Communication Techniques for RDRN 7
3.0 Software Radio Systems 9
3.1 Software Radio Testbed 9
3.2 Network Control with Directional Antennas 10
4.0 RDRN QoS Architecture 11
4.1 The Application Layer 11
4.2 QoS Architecture for RDRN 11
4.3 Flow Specification 11
4.4 Traffic Type 12
4.5 Priority 12
4.6 Traffic Parameters 12
4.7 Quality of Service 12
4.8 Wireless QoS Requirements 13
4.9 Mobile QoS Requirements 13
4.10 Filter Specification 14
4.11 Flow Management 14
4.12 RDRN Flow Management Model 15
4.13 Wireless Multi-Hop Routing Protocol 16
4.14 Flow Management 16
4.15 Flow Establishment 18
4.16 Systems Evaluation 19
4.17 Mobile QoS 22
5.0 System Evaluation 24
5.1 Radio System Performance 24
5.2 IP and ATM Comparison 24
6.0 Conclusion 26
7.0 Bibliography 27
8.0 Appendix – List of Technical Reports 28

1.0 Introduction

1.1 Objective

The primary objective of the RDRN project was to create architectures, protocols, and prototype hardware and software for a high speed network that can be deployed rapidly in areas of military conflicts or civilian disasters where communication infrastructures are lacking and or destroyed (e.g. Desert Storm, Bosnia, Hurricane Andrews, LA, Turkey, Taiwan earth quake).

1.2 Approach

The rapid deployment requirement coupled with higher speed requirements and seamless integration with other commercial networks has lead to an approach that uses wireless technology for the communication links and ATM for networking. The Rapidly Deployable Radio Network (RDRN) being developed by the University of Kansas is a wireless ATM network and it consists of portable (mobile) communication nodes which can be deployed on the ground or on mobile platforms such as trucks, helicopters or fixed wing aircrafts. When deployed, the nodes use GPS derived location information to automatically configure themselves into a high capacity, fault tolerant network infrastructure.

RDRN is made up of two types of nodes: end user nodes providing wireless ATM access for users at a rate of up to 1.5 Mbit/s and edge nodes which serve as Radio Access Points (RAPs) or base stations and provide switching and connectivity between users. Both types of nodes contain GPS receivers for location determination, software controlled radios with phased-array antennas for beam forming and pointing in the right direction using GPS derived location information, and network control software. The edge nodes also have integral ATM (software) switches and they are interconnected by high capacity (45 to 155 Mbit/s) directional radio links. Edge nodes can also interface to wired ATM networks.

The RDRN architecture consists of three overlaid radio networks: (1) a low bandwidth, low power, omni-directional order wire packet radio network for broadcasting location information, network configuration and management; (2) a cellular like ATM radio network for end user access to edge nodes with hand-offs, and (3) a high capacity wireless ATM backbone network providing connection between switches using high capacity radios with multiple directional beams.

When RDRN is initially deployed in a new area, each of the edge nodes initiate the following activities: (1) determine its location from GPS and broadcast it over the secure orderwire network; (2) listen for broadcasts from other nodes; (3) establish the backbone network by forming high capacity, directional radio links to nearby nodes using the steerable phased array antenna; and (4) begin executing the distributed network configuration and control algorithm and establish connectivity with end user nodes. Each edge node is capable of forming multiple radio beams in the direction of other edge nodes or towards end users in the vicinity. A phased array antenna with digital beam forming is used to form these multiple beams, and pointing directions are derived from location information. Assignment of beams to users, node to node connections, and handoffs of users from one edge node to another are controlled by the distributed network configuration and management algorithm. The network control information is broadcast over the orderwire network. Changes in network topology due to mobility or failure of nodes and or links are detected by the network control algorithms and reconfiguration is carried out automatically in a distributed manner.

RDRN is also adaptable to changes in the quality of the radio communication environment. While ATM is designed to operate on high quality (almost error free) wired links, typical radio links suffer higher error rates and the link quality changes as a function of time due to mobility and changes in the environment. By estimating the channel parameters such as multipath spread and signal to noise ratio, communication parameters at the link and network levels are adapted to provide appropriate throughput and quality of service.

This report presents an overview of the results of the RDRN effort, partitioned according to particular technology and focus areas, specifically wireless, radios, networking, and system evaluation. Further more detailed information can be found in the RDRN technical reports cataloged in the Appendix to this report.

2.0 Adaptive Communication Techniques

Mobile radio channels are frequency selective and time varying in nature due to multipath and fading. Mobile communication systems are traditionally engineered to operate at an acceptable BER rate under "average channel conditions" with adequate margin provided over the nominal E_b/N_0 requirements in the link budget to meet outage requirements when the channel goes into a deep fade. An alternate approach is to use an adaptive radio design where the radio communication parameters such as modulation schemes, code rates etc are adjusted to match the time varying channel condition. The basic idea behind this approach is to maximize throughput and quality of transmissions when the channel is in a "good state (no fading)", and reduce transmission rate when to channel is in a "bad state (deep fade)". Implementation of adaptive communication techniques will consist of two major steps:

1. Estimating the instantaneous channel conditions
2. Adapting communication parameters accordingly

The approach studied for channel estimation and adaptive communication in RDRM is summarized below.

2.1 Channel Estimation

The channel estimation problem is one of identifying the impulse response of the channel using a known input sequence and observing the channel output. If A is the channel input vector, h is the channel impulse response at time t, and Y is the channel output vector, then the relevant equations for channel estimation are: $Y = Ah + N; \quad \hat{h} = (A^T A)^{-1} A^T Y$, which can be solved using iterative techniques. Channel estimation can be done with or without reference sequences. While blind estimation algorithms that do not require reference sequences can be used for channel estimation, they are very slow to converge and hence we took the approach that requires the use of a reference sequence.

If the reference sequence is embedded in the payload portion of the ATM cells in the RDRN HDLC frame this will decrease the throughput of the system. Instead, we recommend the use of 12 unused bits in the header portion of each of the three plus ATM cells in the HDLC frame. This gives us 36 bits of reference data per HDLC frame for channel estimation and if we assume the channel to be slowly fading then the estimation can be extended over several HDLC frames.

Simulation results indicate that the impulse response of a multipath channel with a delay spread lasting up to 5 symbols can be estimated accurately with about 200 symbols of reference data (Complex PN sequence). It is possible to improve the accuracy of channel estimation for time varying channels by 50% (reduction in variance of the estimator) using a Kalman filter algorithm.

In the Kalman estimation algorithm, the time varying impulse response of the channel is modeled as an auto-regressive (AR) process with an appropriate Doppler spectrum and the current estimated value of the channel is extended via this model and combined with the next estimate of the channel impulse response. This recursive estimation algorithm is computationally very efficient and can be implemented in real time.

Details of the channel estimation algorithm and simulation results may be found in references [1] and [2].

2.2 Adaptive Communication Techniques for RDRN

Since wireless communication channels are time varying, adaptive communication techniques will be more efficient than communication systems with fixed parameters. If the channel state can be estimated accurately (as described in the preceding section), then it is possible to design an adaptive communication system in which the communication parameters are changed in accordance with the time varying channel characteristics in order to optimize throughput, and performance.

Among the methods investigated for adaptive communications are modulation level controlled systems and rate adaptive systems. The idea behind these two schemes can be summarized as follows:

1. When the channel is in "good" condition, the throughput of the system can be increased by using a higher order modulation scheme while maintaining the same transmit power level and BER.

2. When the channel condition degrades we can maintain a lower throughput while maintaining the same transmit power level and BER by lowering the symbol rate and using a simpler modulation scheme.

The choice of communication parameters will depend on the channel attenuation (i.e. severity of the fading), and also on the multipath spread. Based on extensive simulations, we recommend the following algorithm summarized in Figure 2.1 for the choice of communication parameters based on the estimated channel conditions.

The results shown in Figure 2.1 were obtained by simulating the transmission of date at varying bit rates, modulation schemes, SNR, and delay spreads and estimating the BER. The threshold for the BER was set at $5x10^{-5}$. The channel estimation algorithm estimates the delay spread and SNR for the current frame and based on this the modulation parameters for the next frame are determined The channel is assumed to be slowly varying with time constant of the order of 10 frames.

Details of this adaptive scheme may be found in [3].

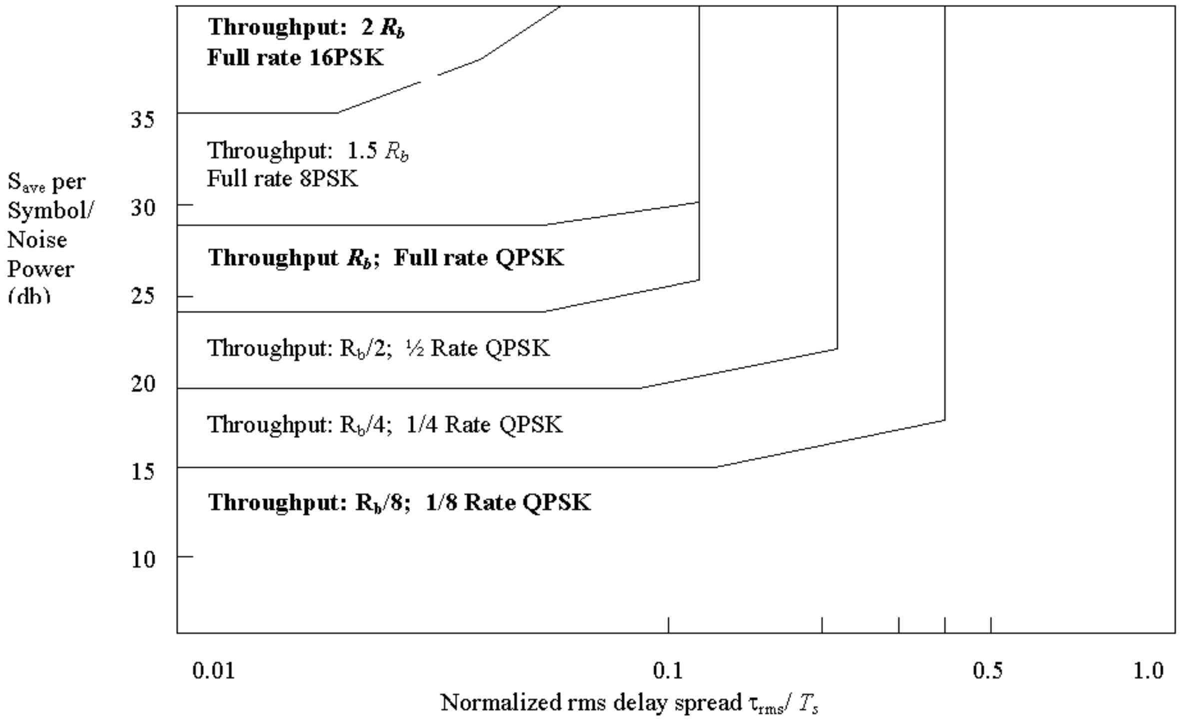


Figure 2.1: Algorithm Results

3.0 Software Radio Systems

Software radio systems have emerged from the research laboratory as a commercially viable and flexible approach for implementing complex radio signal processing functions using digital signal processing technology. Implementing analog signal processing functions using digital hardware offers the potential for reducing the size, weight and power requirements of these systems. Through a combination of software, special purpose microprocessors, digital logic and programmable gate arrays, we implemented a software radio testbed which evolved into fully functional software radio operating in the 5 GHz Unlicensed National Information Infrastructure (UNII) radio band. The testbed was interfaced to a directional antenna array whose active elements are under network control.

3.1 Software Radio Testbed

The software radio testbed was constructed to provide the necessary radio frequency link connectivity between nodes in the Rapidly Deployable Radio Network. A software radio approach was chosen in order to meet requirements for mobility – flexibility, size and power consumption.

The first requirement of any radio system is to downconvert the signal to an intermediate value for demodulation and information extraction. Our system functioned over a grange of frequencies within the 5 GHz UNII band. We placed a low noise amplifier at the antenna and then immediately converted this signal to an intermediate frequency of 70 MHz. At this frequency we were able to capture the bandpass signal by a relatively inexpensive analog-to-digital (A/D) converter.

Modern software radios utilize high-speed A/D converters and digital signal microprocessors to implement many of the radio signal processing functions. The approach is to place a high speed A/D converter as close to the antenna as possible. As a practical matter, high performance receivers need to operate with a low noise figure within some bandwidth. Therefore, software radio A/D converters usually follow a low noise RF filter and amplifier in the intermediate frequency stage of the receiver. Several stages of amplification may be required to raise the input signal level from microvolts to millivolts in order to take advantage of the full dynamic range of the A/D converter.

Once the bandpass signal has been captured, one further downconversion is required to bring the signal to baseband. This is accomplished by the digital downconverter – a system that employs a numerical (digital) oscillator that feeds sine and cosine signals to multipliers. This forms a quadrature detection process that preserves the phase of the signal. The demodulated signal is then converted to binary symbols and sent to the controller for routing.

This system serves as an effective testbed for demonstrating the link-level capabilities and limitations of the network. Bit error rate tests were made in order to determine the individual link throughput and the overall quality of service for the system. An essential component for link-level connectivity is the antenna array and network controller's capability to direct the antenna radiation pattern in any direction.

3.2 Network Control with Directional Antennas

The antenna developed for this project is a multiple element cylindrical array system. Individual patch antennas were attached to a cylindrical structure and a feeder system was developed to split the transmitted energy and route it to the appropriate elements.

The individual array elements are configured so that a digital beamforming system can be implemented under direction of a control system. To validate our concept, a prototype digital beamforming receiver was constructed to confirm that control of the radiated pattern while receiving data was a feasible capability. Independent tests were also conducted to verify that the antenna system was able to effectively radiate multiple beams. The cylindrical array allows 360 degree coverage and multiple beams can be operational simultaneously.

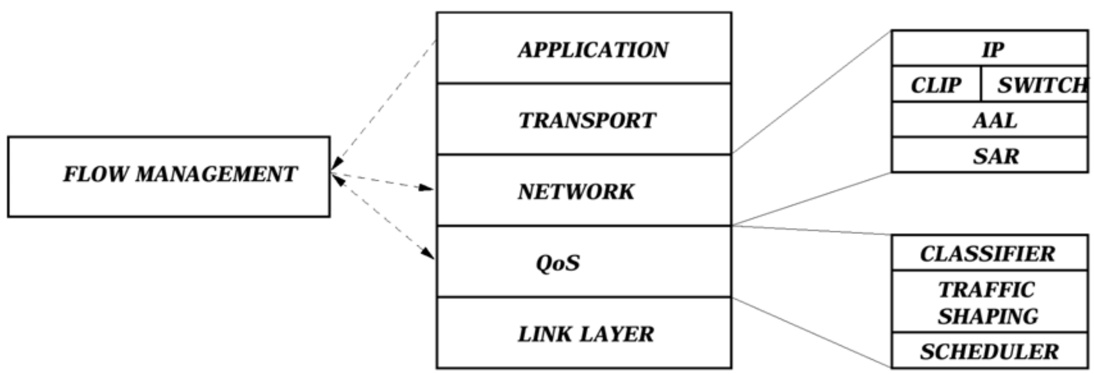
With 360-degree coverage using multiple beams, the RDRN concept becomes a reality. It is possible to link a single RDRN node into a network consisting of one or more other nodes. The combination of the software radio and the directional antenna allows the flexibility needed to establish and control a high-speed data network over radio frequency links.

4.0 RDRN QoS Architecture

Based on the requirements for a robust and mobile return, a QoS architecture has been proposed for Rapidly Deployable management functionality, and its interactions with the rest of the elements in the architecture. This section describes the RDRN QoS architecture in detail.

4.1 The Application Layer

The application layer specifies the following:
1. The *flow specification*, which is used to indicate the required user QoS and specify the characteristics of the traffic that will be generated by the application.
2. The *filter specification*, which is used to identify the packets that belong to the flow.



4.2 QoS Architecture for RDRN

An application programmer interface (API) that can be used to specify the flow specification has been developed. The filter specification is obtained from information available in the packet. The flow specification and the filter specification put together is referred to as the *flow descriptor*. Before going into the details of the rest of the architecture, it is important to first define the flow specification and filter specification that is used in RDRN.

4.3 Flow Specification

Flow specification (commonly referred to as *flowspec*) is a data structure used by the internetwork hosts to request special services. These services often guarantee how the internetwork would handle some of the traffic generated by the hosts. A flow specification (usually referred to as a flow spec) is part of the negotiation that will be done by the host with the internetwork to use a portion of the internetwork's resources. The flow specification is typically used to specify the service requirements in one direction only.

There are two different components in flow specification:
1. The QoS that is requested by the host, which identifies the resources requested by the host to the network.
2. The traffic that will be generated by the host, which identifies the traffic pattern that the network should expect from the host.

This flow specification is tailored for highly dynamic networking environments like the RDRN network. The instability of the wireless links and the highly mobile nature of the nodes influence both the flow specification and the flow specific information that is maintained at each node.

The flow specification that is used by the host for negotiation with the network is discussed next. This specification typically includes the traffic type of the flow, the priority that needs to be assigned to the flow, the traffic parameters and the QoS specification.

4.4 Traffic Type

This represents the type of traffic that the flow will carry and will be either *real time* or *non-real tim*e. The traffic type will determine the type of commitments that will be required from the flow. Real time traffic will need deterministic commitments while non-real time traffic may need only best effort commitments. However, there may be certain non-real time applications that need deterministic commitments too. Such requirements are specified by the application.

4.5 Priority

There may be multiple real time and non-real time applications in the same node (MAP or MN). It may be necessary to give one particular application's traffic preference over another application's traffic, e.g., FTP traffic might need a higher priority over TELNET traffic. For this purpose, the priority field is provided in the flow specification. A class-based queue is used to differentiate traffic from the various applications based on the priority field. Priority can be one of *high priority/ medium priority/ low priorit*y.

4.6 Traffic Parameters

The traffic pattern should indicate the type of traffic that the source expects to give to the network. The traffic pattern is characterized with the help of the token bucket algorithm. The traffic control parameters of interest are the bucket size, maximum transmission rate, maximum burst size and token arrival rate.

4.7 Quality of Service

The QoS field should indicate the quality of service that the host expects from the network, given the traffic that will be generated by the host. For real-time flows, there are essentially two types of payloads, namely the *base layer* and the *enhancement layer*. The requirements from both these layers need to be specified in the *flowspec*. The base layer represents the minimum QoS requirements of the flow while the enhancement layer represents the maximum QoS requirements, both of which are specified in the flowspec.

The end-to-end QoS will characterize the traditional QoS parameters supported in wired networks, namely the *delay, jitter, loss and throughput*. This typically represents the QoS that will be directly requested by the user, and that which is directly visible to him. The delay indicates the expected *end-to-end delay*. The jitter indicates the *end-to-end delay variation*. The loss indicates the *loss ratio* that is acceptable. The throughput indicates the *number of packets* that are received successfully without any error. As mentioned earlier, the media type will be used to determine the end-to-end QoS parameters. For example, in the case of audio, there is a stringent requirement in terms of the end-to-end delay and jitter. Throughput and reliability can be compromised to a certain extent as far as audio is concerned. In general, for audio, the throughput requirement can be as low as 4 Kbytes/second, the end-to-end delay requirement is 100ms and jitter acceptable is 10ms. These values are derived from the specifications given by the end user. Usually, for non-real time traffic the throughput is very important while the delay is not a very significant parameter. Derived Flow Specification This section discusses the flow specific information that is maintained by the mobile access points in the RDRN system. The flow specific information is derived from the flow specification that is sent by the host. Based on the nature of the RDRN system, the derived flow specific information that is maintained in the nodes consists of two aspects, namely the *wireless QoS* and the *mobile QoS* requirements.

4.8 Wireless QoS Requirements

The wireless QoS parameters that need to be maintained in the derived flow specific information deals with the nature of the wireless links. This includes the *link delay, error rate* and the *channel reservation*.

The *error rate* is derived based on the media type and the required end-to-end loss. For non-real time traffic, the error rate has to be very low, whereas for real-time traffic, the error rate can be higher. Additional protection can be offered to the packets desiring a low error rate, at the link level. The *link delay* is derived based on the media type and the required end-to-end delay. For real-time traffic, the link delay has to be very low, and it also depends on the required end-to-end delay. For non-real time traffic, though, the link delay is not a very significant parameter, and the delay need not be guaranteed very strictly. The *channel reservation* is influenced by the priority that is specified in the flow specification. For high priority flows, a higher channel bandwidth is reserved. These are the QoS requirements that arise because of the characteristics of the wireless links.

4.9 Mobile QoS Requirements

Mobile QoS is mainly concerned with the QoS associated with the handoff. Each flow is associated with certain handoff parameters, namely the *handoff urgency* and the *handoff* *loss.* This flow specific information that is maintained in every node is also derived from the flow specification specified by the host application.

The need to do a seamless handoff leads to *the handoff urgency* or *deadline parameter,* which represents the priority that needs to be assigned to this handoff process. This could be one of *fast, medium or slow* representing the quickness with which the handoff needs to be completed. This is derived from the media type and the required delay. Real time flows typically need low end-to-end delay and as a result, need a fast handoff. Non-real flows vary widely in delay requirements, and might require either a medium or a slow handoff.

The *handoff loss* (i.e. no loss allowed, loss allowed) essentially determines the type of handoff that needs to be done and is again dependent on the media type of the flow and the end to end loss requirements specified in the flow specification. Typically, for real-time flows, fast handoff takes precedence over the loss. As a result, the urgency is set to a fast, whereas the loss is set to *loss_allowed.* For non-real time flows, the urgency is set to slow, while the loss is set to *no_los*s. These are the QoS requirements that arise because of the highly dynamic nature of the RDRN environment.

4.10 Filter Specification

The filter specification (commonly referred to as *filterspe*c) provides the identification for the flow that is to get the QoS specified by the *flowspe*c. The filter specification consists of the source IP address, the destination IP address, the source port number, the destination port number and the protocol (UDP/TCP). This information is available in all the packets that are sent from the source to the destination. Thus the filter spec is identified by the 5 tuple:

<*Source IP, Source Port, Destination IP, Destination port, Protocol*>

The next section discusses the flow management block, which is the main functional unit in the QoS architecture.

4.11 Flow Management

This section describes the flow management process in the RDRN QoS architecture. Flow management is the most important functional block in the architecture. The detailed flow management block is shown in Figure 4.1. It is responsible for performing the following functions:

1. *QoS mapping:* The application layer specifies the type of the application i.e. *real-time* or *non-real tim*e, as part of the *flow specificatio*n. The flow management block maps the flow specification into a type-of-service (TOS) byte. The TOS byte is interpreted by the intermediate nodes for the provisioning of specific services to the flow. The application specifies its requirements in the form of a flow specification, which is mapped by the flow management module to a TOS byte. This is given as an input to

the classifier block in the architecture, which classifies the data cells to the appropriate queues.

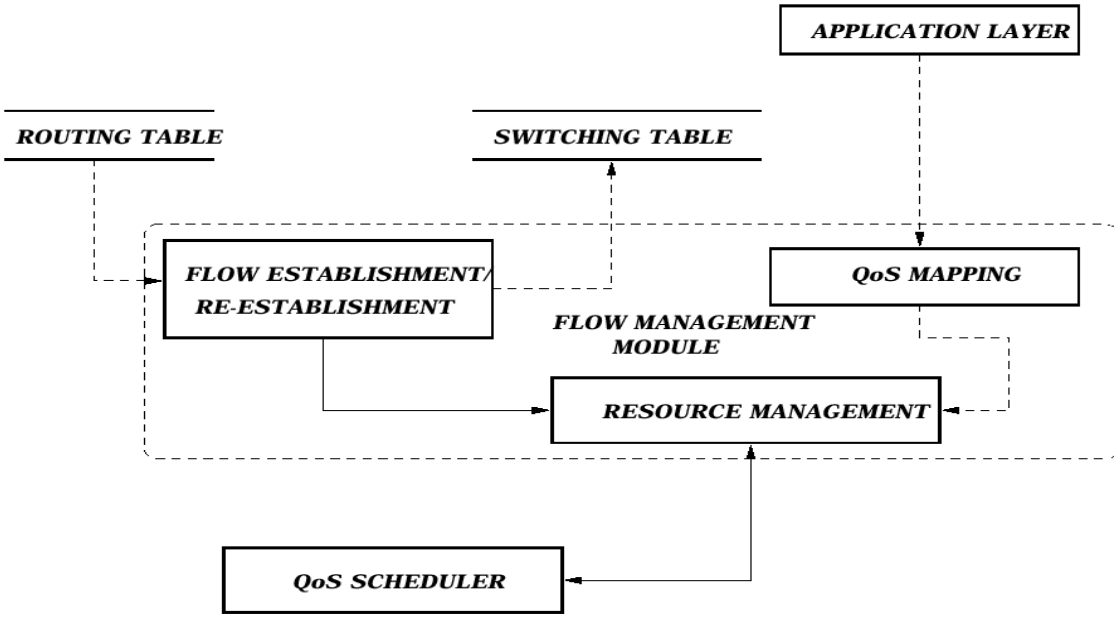


Figure 4.1: Flow Management Module

2. *Flow Establishment/Re-Establishment:* The flow management block is responsible for end-to-end flow establishment. An in-band flow establishment protocol has been developed for this purpose. The flow establishment protocol will set up end-to-end flows based on the QoS requirements of the flow. The flow management block is also highly responsive to changes in the network (e.g., failure of a wireless link) and re-establishes the flows from the source to the destination. As part of this functionality, the flow management block controls the switch and the Classical IP over ATM (CLIP) blocks.

3. *Resource Management:* The flow management block is also responsible for resource management in the RDRN network. It does flow admission control to determine if a flow's requirements' can be satisfied. The intermediate nodes use the TOS byte to interpret the requirements of the flow. It then configures the scheduler for the resources requested.

Before going into further details on the various functions of the flow management block, it is important to understand the overall flow management model, that is, the various components that will be involved in flow management.

4.12 RDRN Flow Management Model

This section discusses the RDRN flow management model. The goal of the RDRN Flow Management model is to support delivery of real-time and non-real time services to the

RDRN nodes under time-varying conditions. The flow management model allows packet audio, video and real-time data application to specify their maximum and minimum QoS requirements using the flow specification discussed in the previous section. The RDRN flow management model is shown in Figure 4.2.

4.13 Wireless Multi-Hop Routing Protocol

This protocol dynamically tracks changes in the RDRN network topology and makes the current routing table visible to flow management module. The wireless multi-hop routing protocol maintains multiple paths to the destination node and installs the best route among these in the routing table. It also ensures a loop-free path to the destination with the help of the predecessor information that is maintained for each route. The routing protocol exchanges HELLO packets with the nearby nodes to indicate that the node and link are active. Absence of three consecutive HELLO packets is interpreted as a link failure and an alternate route is installed for the destination. The flow management module will use the routing table for the flow establishment and the flow re-establishment process.

4.14 Flow Management

As already mentioned in the previous section, the flow management block is responsible for the end-to-end flow establishment, flow re-establishment (in the event of a change in the path from the source to the destination) and resource management. The flow establishment protocol, as will be seen later, uses the routing protocol in the event of a link failure. Also, it will set up the switching table entry for a flow during the process of flow establishment. The flow management block also sets up the scheduler for the transmission of cells.

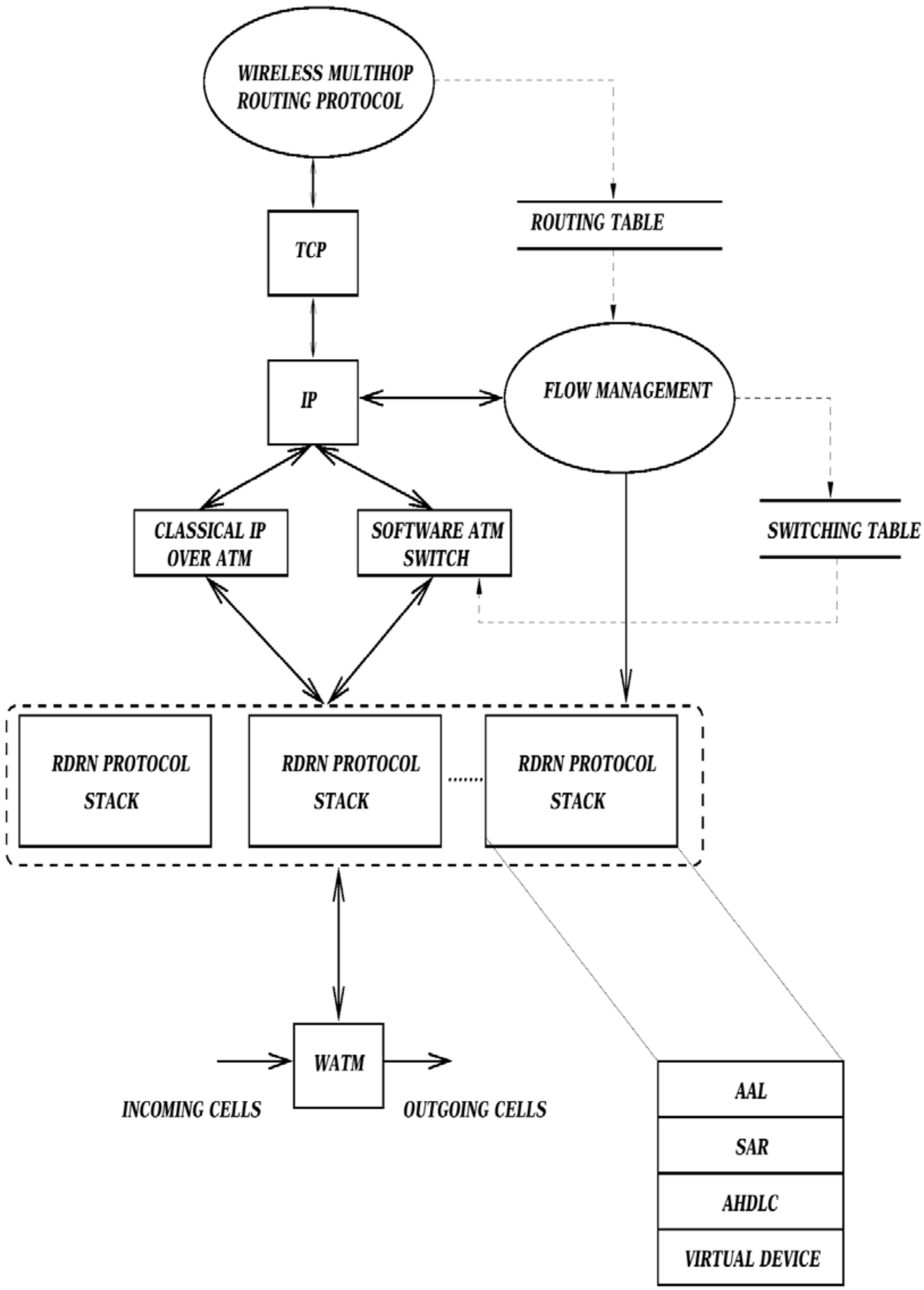


Figure 4.2: RDRN Flow Management Model

4.15 Flow Establishment

Flow establishment is the process of setting up the flow from the source to the destination. This involves reservation of resources in the nodes from the source to the destination. The RDRN system has a highly dynamic networking environment. The RDRN protocol stack consists of IP over ATM. It is different from typical mobile adhoc networks in the sense that hierarchy is imposed on the network by the presence of ATM in the protocol stack. The proposed flow establishment scheme intends to make use of the features of both IP and ATM in the protocol stack. The flow establishment protocol uses the routing table that is built by the wireless multi-hop routing protocol discussed in the previous section.

In the RDRN system, the MAPs discover each other (using the low speed orderwire system) and set up high speed point-to-point wireless ATM links to each other. A default ATM VPI/VCI is used for the exchange of IP datagrams between the MAPs. The default VPI/VCI is used for all the flows through the MAP for which a specific VPI/VCI has not yet been allocated. Specific VPI/VCIs will be set up during the process of flow establishment, and will be used once the end-to-end flow has been established. This default VPI/VCI is also used in the process of flow re-establishment.

The end-to-end flow establishment is done at the IP layer and is accomplished through the introduction of a new IP option field. This IP option is defined as the *QoS option.* The format of the QoS option is shown in Figure 4.3.

REQ/RES	***MAX/MIN/BEST***	***ALLOC/DE-ALLOC***	***VPI***	***VCI***
1	***2***	***1***	***16***	***16***

Figure 4.3: QoS Option Field

The basic idea of the proposed flow establishment scheme is to establish the flows at the IP layer. Once the flow is set up at the IP layer, the data is then switched at the link level (ATM). Switching at ATM layer is faster because the datagrams need not be reassembled at every MAP in the path from the source to the destination. The IP datagrams can be reassembled at the destination. Also, the time involved in processing the QoS option at every node can be avoided if layer 2 switching is done.

The establishment of the flows is done in-band, i.e., along with the transfer of data. Since the resources in a wireless environment are scarce, in-band signaling serves to improve the efficiency by avoiding the additional overhead in out-of-band signaling. The MAPs all maintain soft state, and as a result, the resources are released in the absence of data on the links. See RDRN document.

4.16 Systems Evaluation

The performance of the RDRN QoS architecture was evaluated using the RDRN emulation environment. To test the validity and determine the performance of the proposed QoS architecture for RDRN a set of metrics was developed. Broadly, there are two entities that need to be tested in the architecture. The first entity is the flow management module, while the second entity is the QoS layer in the revised RDRN protocol stack. From the flow management module's perspective, the metrics include:

1. *Flow Establishment tim*e: This will be the time taken to establish the end-to-end flow. The time interval between the transmission of the first packet and the receipt of the QoS Report message from the destination is referred to as the flow establishment time. This parameter will be measured for various numbers of Mobile Access Points from the source to the destination.

2. *Flow Re-establishment tim*e: This will be the time taken to establish the end-to-end flow from the source to the destination, in the event of an intermediate node losing connectivity to the next node in the path. This again will be measured for various numbers of Mobile Access Points from the source to the destination.

3. *End-to-end throughput with/without flow-establishment with/without mobilit*y: This set of experiments will be used in determining the efficiency of the proposed flow establishment protocol. End-to-end flows are set up using the flow establishment scheme, and the throughput is measured and compared with IP level forwarding of the packets. Again, the same set of experiments will be repeated with mobility, and the throughput is measured.

This section discusses the experiments conducted to test the scalability of the proposed flow establishment protocol. The logical configuration of the testbed is shown in Figure 4.4. Flows are set up from testbed A to testbed Z via the 10 testbeds and the flow establishment time is measured. The end-to-end throughput is also determined. The experiments are performed multiple times. The mean and the deviation from the mean of the flow establishment time are plotted in Figure 4.5. This plot also shows the flow establishment times for different number of MAPs between the source and the destination. The results show that the flow establishment times increase with an increase in the number of MAPs between the source and the destination. This increase is reasonable, indicating one facet of the scalability feature. The throughput obtained has been plotted in Figure 4.6. The plot shows a marginal decrease in the throughput with an increase in the number of MAPs between the source and the destination. This indicates the other facet of the scalability of the architecture.

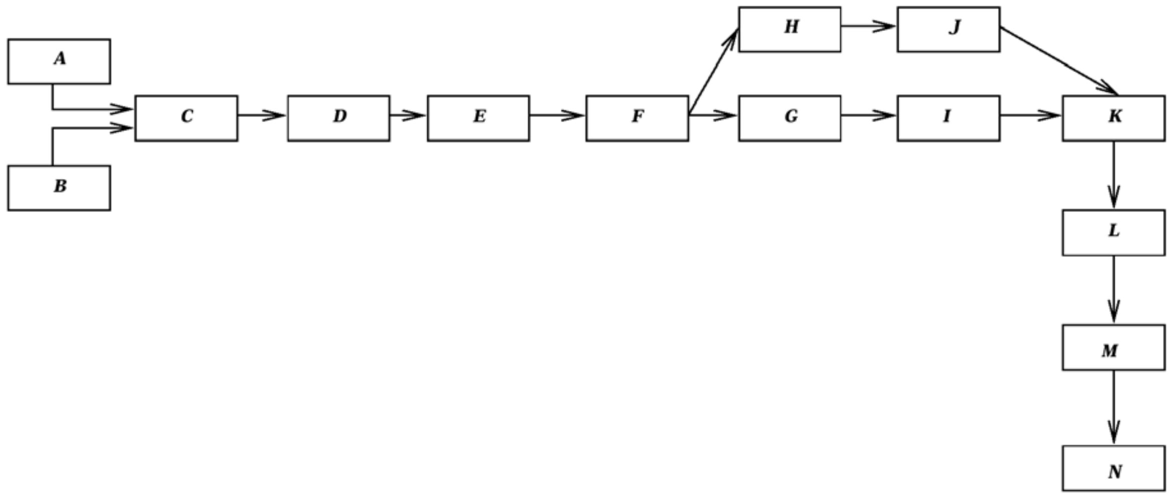


Figure 4.4: Logical Test Configuration to Test Scalability

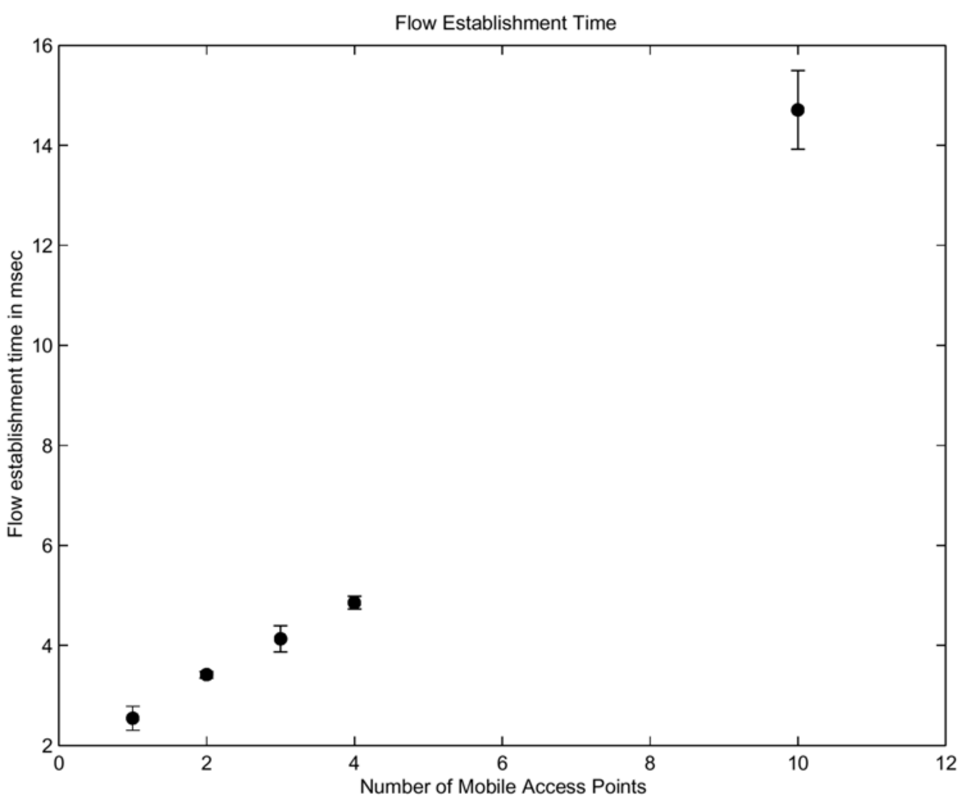


Figure 4.5: Scalability – Flow Establishment Time

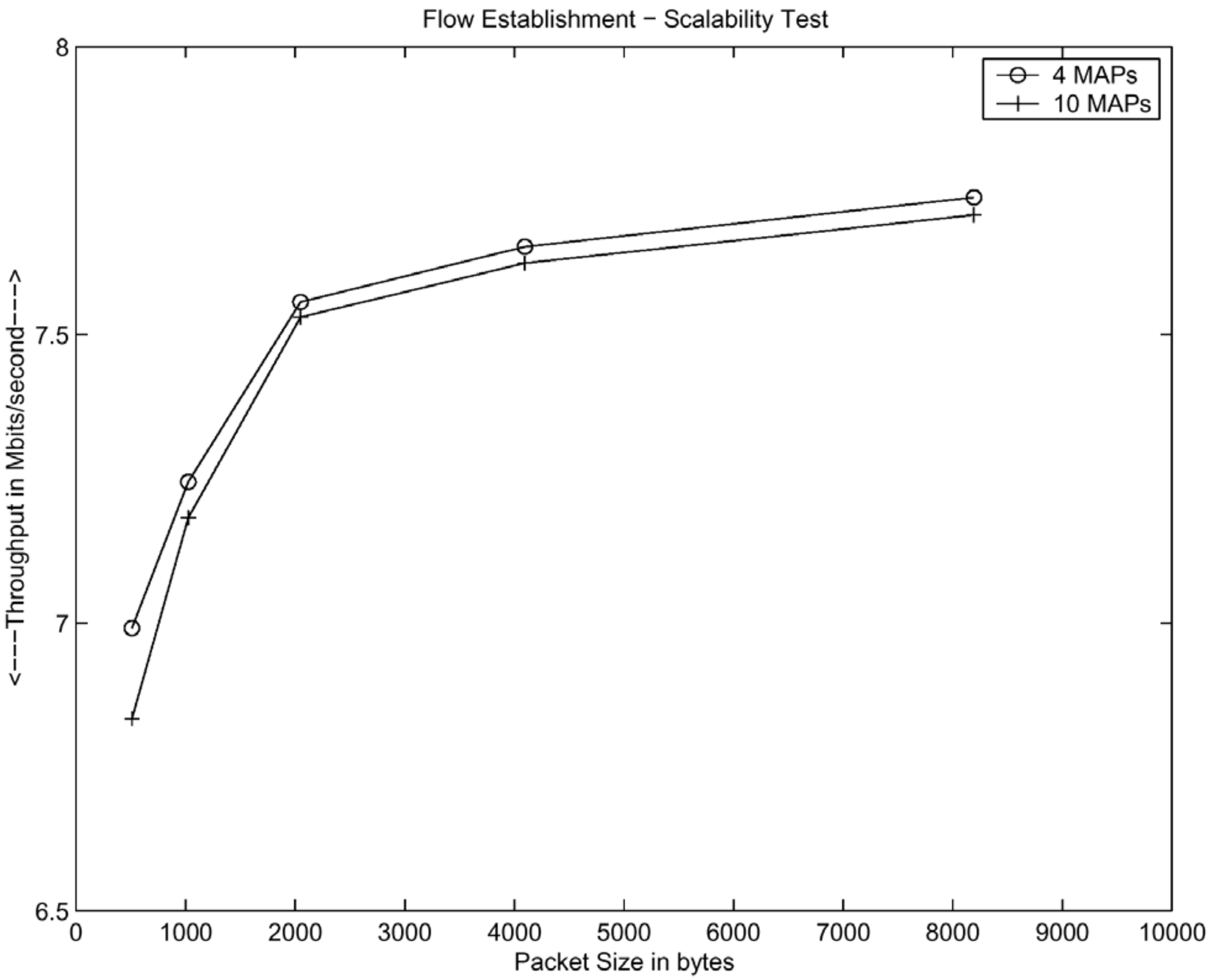


Figure 4.6: Scalability – Throughput

4.17 Mobile QoS

This section discusses the experiments that were conducted to test the validity of the QoS layer in the presence of highly dynamic networking conditions. The emulation manager, developed at the University of Kansas, was used to emulate moving nodes. The scenario that was used to test this feature is shown in Figure 4.7. This scenario consists of four MAPs and 3 MNs.

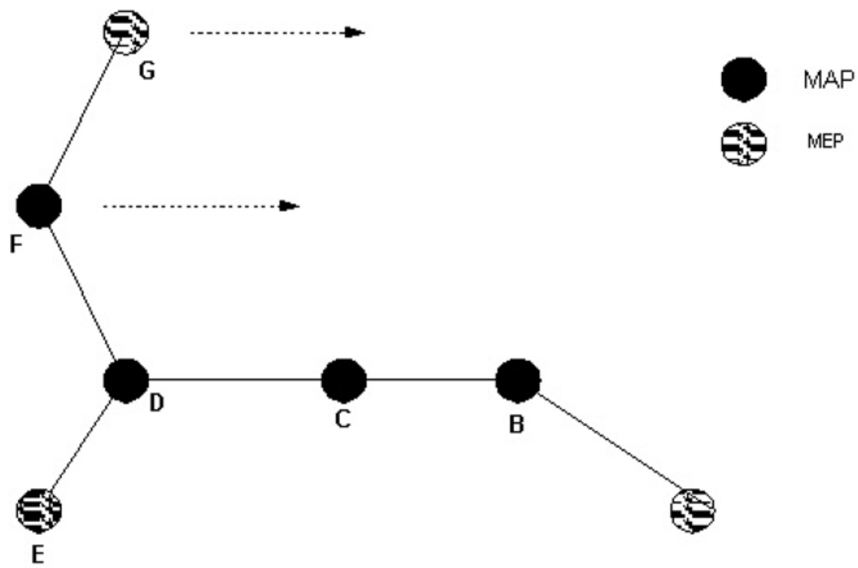


Figure 4.7: Scenario to Test Mobile QoS

As shown in the figure, MAP F and MN G, keep moving eastwards at the same speed. MAP F serves as the only point of attachment for MN G. As the nodes keep moving, a flow is established from MN E to MN G. Initially the path used would be via MAPs D and F. After sometime, the link between MAPs F and D breaks, and the flow is re-established from E to F via MAPs D, C and F. Again, after sometime, the link between MAPs C and F breaks, and the end-to-end flow is re-established via MAPs D, C, B and F. The end-to-end throughput is measured and the results are tabulated in Table 4.1. The results show that the rate at which the packets are sent at the source is almost equal to the rate at which the packets are received at the destination, inspite of the dynamic condition in the network. The test was then repeated for two flows from the source to the destination. One was a high priority flow and the other was a low priority flow (in the ratio of 5:2). The results are tabulated in Table 4.2. The results show that even as the high priority and the low priority flows send data at the same rate, because of the ratio of the weights, the throughput achieved by the high priority flow is higher than that achieved by a low priority flow, thereby proving that the QoS characteristics are maintained even under highly dynamic conditions.

Table 4.1: Throughput achieved by an end-to-end flow in a highly dynamic environment

Sending Rate (in KB/second)	**Receiving Rate** (in KB/second)
466.17	466.15
459.61	459.59
464.19	464.17
462.25	462.02
467.27	478.39

Table 4.2: Throughput achieved by two flows simultaneously in a highly dynamic environment

Flow 1		**Flow 2**	
Sending Rate (in KB/second)	**Receiving Rate** (in KB/second)	**Sending Rate** (in KB/second)	**Receiving Rate** (in KB/second)
483.34	472.64	466.16	393.60
479.83	467.12	459.59	396.80
483.41	470.74	464.79	392.80
481.13	474.33	457.84	394.53

5.0 System Evaluation

5.1 Radio System Performance

The performance of the RDRN radio systems was evaluated in several field tests, as well as demonstrated in complete system demonstrations at DARPA events.

One of our typical field tests had the goals of evaluating performance of the radios and comparing result against data collected in laboratory; running typical applications such as Microsoft NetMeeting across the wireless link and evaluating performance, and running FTP and evaluating link speed and utilization. In order to accomplish these goals, we performed a one-way (up or down link) bit-error-rate test, a two-way (up and down link) loop-back bit-error-rate test, and then an application test with ping, Microsoft NetMeeting, and FTP.

This typical field test was very successful. Transmit power (ERP) was ~20dBm(100mW) when the distance between the two nodes was ~4 miles (~6.5km). The radio achieved a bit-error-rate(BER) of 3e-7 with Eb/No of around 16dB, which is very close to the performance obtained in a "wired" link in the laboratory. NetMeeting ran well with no major picture quality defect, except when there were cars passing by in front of the radio at the test site. The FTP transfer rate was 66.49kBps with NetMeeting running. With a 1 Mbps wireless link, FTP utilized almost 50% of the link capacity while NetMeeting was still running in the background with no noticeable picture quality degradation (note the link is not completely error-free). The bit-error-rate on the loop-back test was 6e-7, which verified the bit-error-rate of a one-way link with half the bit errors.

5.2 IP and ATM Comparison

The rapidly deployable network is an integrated mobile networking system. It supports mobile switching nodes, interconnected over point to point radio frequency links, and supports applications with both Internet protocol (IP) and asynchronous transfer mode (ATM) communication services. The RDRN system can also interoperate with wired infrastructure over fiber optic links and satellite systems over radio frequency links. RDRN is a self-organizing mobile network.

When RDRN was first proposed, the Internet protocol (IP) offered ubiquitous interconnectivity between a multitude of applications. Asynchronous Transfer Mode (ATM) offered the ability to shape and control traffic flows in a low bandwidth wireless network. Hence, RDRN combined both ATM and IP in the switching nodes; ATM providing traffic shaping and IP providing interconnectivity. An RDRN switching node contained a software ATM switch, multiple virtual ATM devices and perhaps some real ATM devices. The real ATM devices would interconnect to fiber optic networks. The virtual ATM devices would connect to wireless links. As ATM cells arrived at a switching node, they could either be switched through the software ATM switch and rerouted back out a different port or be collected into a IP package and forwarded to the

IP processing layer. In the IP processing layer, packets could either be routed through a normal IP routing mechanism or collected and sent to a higher level such as UDP, TCP and other applications.

Using this combination of ATM and IP seemed a reasonable idea. ATM provided traffic shaping and quick access to link capacity due to the small cell size and IP provided interoperability with numerous operations. However, questions continuously arose as to the efficiency of ATM, particularly over low bandwidth wireless links. There were additional concerns about the delays induced if only larger IP packets were sent over the RF links. We performed efficiency and performance studies in order to understand some of the trade-offs between using ATM and IP protocols and some recommendations for future mobile networking systems.

To summarize, ATM cells offer lower average queuing delay to high priority traffic. For short messages, ATM is only slightly less efficient than IP due to the shorter ATM header. Hence, for shorter messages the shorter ATM headers increase efficiency. For longer packets, there is an increased delay variance through multihop network as high priority traffic must wait at each node for the packet currently on the link.

However, conventional ATM signaling and state maintenance is too complex for a highly mobile network. Circuit set up requires processing at each hop and end-to-end communications. It is extremely difficult to maintain circuit state in mobile switching nodes and the interactions with the Internet Protocol through Classical IP (SEAL IP) and ATMARP are complex.

Our recommendation is to consider radio frequency based mobile networks to use a small fixed packet size or small segment size, perhaps with a circuit identifier at the beginning similar to that used in SLIP and PPP. Future work should be invested in exploring methodologies for rapid flow setup and state maintenance, perhaps along the lines of MPLS.

6.0 Conclusion

This report has provided an overview of the Rapidly Deployable Radio Network (RDRN) project, which has developed architectures, protocols, and prototype hardware and software for a high-speed network that can be deployed rapidly in areas of military conflicts or civilian disasters where communication infrastructures are lacking and or destroyed. The RDRN system uses wireless technology for the communication links and ATM for networking. When deployed, the nodes use GPS derived location information to automatically configure themselves into a high capacity, fault tolerant network infrastructure. This report has presented an overview of the results of the RDRN effort.

Further more detailed information can be found in the RDRN technical reports cataloged in the Appendix to this report.

7.0 Bibliography

[1] P. Rajagopalan, and K. Sam Shanmugan, “Channel Estimation for RDRN Using HDLC Frames”, TISL Technical Report, June 1997.

[2] K. Sam Shanmugan, Channel Estimation Using Kalman Algorithm”, Proceeding of the International Personal Wireless Communications, Hydrabad, India, December 2000.

[3] P. Rajagopalan and K. Sam Shanmugan, “Channel Estimation and Adaptive Communication Techniques for RDRN”, TISL Technical Report, January 1997.

8.0 Appendix – List of Technical Reports

ITTC-FY98-TR-10920-28
Rapidly Deployable Radio Network, Final Report
K. Sam Shanmugan, Gary Minden,, Victor S. Frost, Joseph B. Evans, Glenn Prescott, Richard Plumb, David Petr and James Roberts
Dercember 1997

ITTC-FY99-TR-13380-01
Design of the RDRN Wireless Communications Link: A New Methodology Based on Channel Measurements
Timothy Galligher and James Roberts
July 1998

ITTC-FY99-TR-13380-02
Decision Weighted Adaptive Algorithms with Applications to Wireless Channel Estimation
Shane M. Haas, Glenn E. Prescott, Joseph B. Evans and David W. Petr

ITTC-FY99-TR-13380-03
Resource Reservation and Flow Establishment in rapidly Deployable Radio
Saravanan Radhakrishnan, Joseph B. Evans and Victor S. Frost
April 1999

ITTC-FY99-TR-13380-04
Flow Specification for Rapidly Deployable Radio
Saravanan Radhakrishnan, Joseph B. Evans and Victor S. Frost
April 1999

ITTC-FY99-TR-13380-05
Design and Implementation of a Link Level Adaptive Software Radio
Richard A. Killoy and Joseph Evans
July 1999

ITTC-FY99-TR-13380-06
Multi-Path Routing Protocol for Rapidly Deployable Radio Networks
Fadi Wahhab, Gary Minden and Joseph Evans
February 1999

ITTC-FY99-TR-13380-07
In-Band Flow Establishment for End-to-End QoS in Rapidly Deployable Radio Networks
Saravanan Radhakrishnan, Victor Frost and Joseph Evans
August 1999

ITTC-FY99-TR-13380-08
Emulation of RDRN on an ATM-Testbed and a Comparative Evaluation of IP vs ATM
Syed Fazal Ahmad, Gary Minden and Joseph Evans
September 1999

ITTC-FY99-TR-13380-09
A Multipath Chennel Estimation Algorithm Using a Kalman Filter
Rupul Safaya, Sam Shanmugan and Joseph Evans
July 2000

ITTC-FY99-TR-13380-10
Rapidly Deployable Radio Network Network Control of Sectorized Antenna Array
Artur Leung and Joseph Evans
September 2000

ITTC-FY99-TR-13380-11
RDRN2 System Integration and Testing
Daniel Depardo and Joseph Evans
September 2000

ITTC-FY99-TR-13380-12
RDRN Emulation Manager
Leon Searl and Joseph Evans
September 2000

ITTC-FY2001-TR-13380-13
RDRN Network Performance Studies
Pooja Wagh, Karthik Thyagarajan, Gary Minden and Joseph Evans

ITTC-FY2003-TR-13380-14
Rapidly Deployable Radio Network ATM/IP Analysis
Gary J. Minden, Joseph B. Evans

ITTC-FY2003-TR-13380-15
Rapidly Deployable Radio Network – Phase 2 Final Report
Joseph B. Evans, K. Sam Shanmugan, Gary J. Minden, Victor S. Frost, Glenn Prescott