OVERLAY MULTICAST FOR REAL-TIME DISTRIBUTED SIMULATION

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LIST OF ABBREVIATIONS/SYMBOLS

ACE - Application Characterization Environment
AS - Autonomous System
ASM - Any Source Multicast
ATM - Asynchronous Transfer Mode
C3I - Command, Control Communications and Intelligence
COI - Community of Interest
DoD - Department of Defense
DoDAF - Department of Defense Architecture Framework
DiffServ - Differentiated Services
DMSO - Defense Modeling and Simulation Office
DREN - Defense Research and Engineering Network
DVMRP - Distance Vector Multicast Routing Protocol
GMU - George Mason University
IP - Internet Protocol
ISP - Internet Service Provider
LAN{s} - Local Area Network(s)
MMPP - Markov modulated Poisson Process
MPLS - Multiprotocol Label Switching
MST - Minimum-Weight Spanning Tree
NETLAB - Network Modeling and Simulation Laboratory at George Mason University
ODU - Old Dominion University
OMNI - Overlay Multicast Network Infrastructure
OPS-SIM - Operational Simulation
OV-1 – High-Level Operational Concept Graphic defined by DoDAF
OV-2 – Operational Node Connectivity Description defined by DoDAF
P2P - Peer-to-Peer
PC - Personal Computer
PIM-SM - Protocol Independent Multicast – Sparse Mode
QoS - Quality of Service
RSVP - Resource Reservation Protocol
RT-DVS - Real-Time Distributed Virtual Simulation
RTI - Run Time Infrastructure
RTT - Round Trip Delay
SISO - Simulation Standards Interoperability Organization
SRMP - Selectively Reliable Multicast Protocol
SV-4 – System Functional Description as defined by the DoDAF
TAG - Topology Aware Grouping

VMASC - Virginia Modeling, Analysis and Simulation Center
WSIM - Web Service Interest Management
XML - Extensible Markup Language
XOM - Extensible Modeling and Simulation Framework Overlay Multicast
XOMR - Extensible Modeling and Simulation Framework Overlay Multicast Relay

ABSTRACT

OVERLAY MULTICAST FOR REAL-TIME DISTRIBUTED SIMULATION

This report provides a top-level architecture for an overlay multicast service in support of distributed real-time virtual simulations over an open network environment. The top-level architecture for the overlay protocol was developed based on key concepts identified in the laboratory performance studies, the analytical model, and the studies of live simulations. The proposed protocol is called the Extensible Modeling and Simulation Framework Overlay Multicast Relay Protocol (XOMR). The XOMR provides overlay multicast message relay services to support many-to-many communications in the very dynamic environment of distributed real-time virtual simulation. (See Appendix B for definition of overlay multicast.)

Appendix A includes results from demonstration of the prototype protocol based on the top-level architecture. The results demonstrate that overlay multicast service is a viable mechanism to meet the demands of communications messaging for distributed real-time virtual simulation. A message traffic generator and test environment was developed for use in performance evaluation of a prototype protocol used to demonstrate feasibility for use of overlay multicast to support distributed real-time simulation.

Appendix A also includes a description of an analytical model developed to characterize the message flow characteristics to determine the regions of feasibility for performance for overlay multicasting to support the target application environment. The analytical model was validated in laboratory measurements of the prototype and the results used to influence the design of the overlay protocol architecture.

Results of studies conducted of live real-time distributed simulations operating in the laboratory and over the Defense Research and Engineering Network (DREN) are also included in Appendix A. Included are results of studies and analysis of simulations in the laboratory used to help characterize the operating system performance that might be required to host the overlay protocol.

XOM TOP-LEVEL ARCHITECTURE

I Introduction

This report describes a top-level architecture for the XOM that recognizes that underlying networks may have a wide range of network capacities and capabilities and do not necessarily offer a multicast service. The proposed architecture provides a multicast service to higher layer applications that require this capability across open networks. The approach includes consideration for reliability by providing two classes of services on top of existing UPD/IP protocols and security through a central registry service and implementation of specific features within the protocol to support more secure environments when required.

The research work described in Appendix A laid the ground work and describes the basic principles for defining this architecture for the XOM to support many-to-many multicast for RT-DVS applications. These applications have great demands for network message throughput required for communicating object status updates [Brut02]. This may consist of thousands of updates to simulation objects. A receiver set could be required to support up to 10,000 simultaneous objects per group, and if the capability exists demand could grow to scale up to millions. These object updates typically have message sizes in the range of 100 to 200 bytes without tag or other protocol overheads. RT-DVS are run on heterogeneous set of workstations with differing processing and display capabilities,

traveling over a heterogeneous network with capacities varying by many orders of magnitude between the initial down link and the slowest end user.

The overlay multicast middleware is defined as the XOM Relay (XOMR) where relay implies forwarding or routing of messages to designated destinations from authorized sources. XOM uses an overlay multicast protocol designed to support efficient, reliable many-to-many multicast transmissions on top of existing network protocols such as UDP/IP for real time distributed visual simulations. It is based on the notion of a single multicast host per subnet which controls all aspects of communications on a set of multicast groups as a service to supported applications on that host's subnet.

The simplest syntax definition for the protocol is that a message *m* is sent by process *p* and the reception of *m* is by process *q* at one to many recipients. In order to add a level of QoS, by queuing algorithm, the XOMR provides for the arriving of *m* at an incoming channel from the application interface in an order which is a function of the priority of *m* and provides for queuing on the sending side to the network, by process *p*.

The Department of Defense (DoD) Architecture Framework (DoDAF), Version 1.0 is used as guide for the description and presentation of the top-level architecture for the XOM. The Architecture begins with a discussion of overall design goals of the XOM followed by an operational view in paragraph III. Paragraph IV provides system and technical details of the top level architecture for the overlay multicast protocol. Paragraph V follows with a proposed top-level description of the XOMR routing protocol.

II XOM Design Goals

The key design goals for the XOM are:

- XOM should not require support from underlying network routers or operating systems in order to preserve ubiquitous deployment.
- XOM should be compatible with evolving IP and MPLS multicast services as they are deployed in the Internet and be able to use these services automatically in the underlying networks as they become available.
- XOM uses a central registry service and the location of the registry service should not impact overall performance of the XOM.
- XOM will be self organizing in the sense that configuration is limited to selecting a registry.
- XOM will be compatible/interoperable with existing IP multicast systems, thereby preserving the high value invested in this approach within the installed base.
- XOM will include security features to hence protection of multicast denial of service attacks and provide information channel security when desired.

Good protocol design practices dictate the XOM:

- Use layered design to indicate the logical structure of the XOM protocol by separating tasks. This defines the problem, the service to be performed at every layer, the external functionality, and the internal functionality.
- Use routing middleware that is both scalable and decentralized, e.g., not dependent on a central or root services for routing functionality.

- Is based on standards and portable abstractions of the system with network-specific advantages including scalability, fault tolerance, and resource availability easily utilized without any concerns about their underlying infrastructure and resources.
- Transmission errors are handled at a higher layer (e.g. using the Selectively Reliable Multicast Protocol (SRMP) [Pull99, Moen01] or at the application layer).
- Use system aware messaging so that changes in system status can proactively result in network/application adaptation.
- Is able to deliver/manage QoS to multiple simulation/applications.
- Use application knowledge of DVS to tailor design and implementation.
- Use local algorithms to collectively achieve a desired global effect. Example: Some form of explicit congestion notification could be used for dynamically regulating admitted real-time sessions in the face of network congestion/network dynamics, to take advantage of the network awareness characteristic of RT-DVS applications.
- Translate application level performance requirements into network performance requirements.

Best practices for software system design calls for middleware that:

- Is light in computation and communication requirements.
- Is designed to intelligently trade the QoS of various demands against each other.

- Identifies optimization metrics for use in resource allocation
- Is designed such that changes in topology and network conditions, even node/link failures should not affect the operation of the control mechanism.
- Is designed not to keep per flow or aggregate state information, in order to not have complex signaling for per flow state information as is the case might be with "stateful" QoS approaches.
- Be administrator – friendly. Acceptance depends in part on the willingness of administrators to deploy it, and for the software to be a good network citizen that does not violate corporate security standards or take unfair advantage of protocols that reduce information flow in order to control network congestion.

To maintain topology control of the overly the XOM must have ability to:

- Discovering neighbors
- Identifying position
- Determining transmission radius (diameter of the overlay)
- Establishing links to neighbors
- Maintain selected structure and information about nodes
- Information about service/node access (capacity)

III XOM High-Level Operational Concept

Many-to-many multicast transmission is an essential network capability for scalable distributed simulation because the more common unicast approach does not scale. Here we define many-to-many multicast to mean that many senders simultaneously can send to

all of many receivers. This also is called Any Source Multicast (ASM). Providing robust multicast services to real time distributed virtual simulation (RT-DVS) is an important requirement to enable use of Web based services across open networks such as the Internet. These services must include network level quality of service (QoS) for reliability and bounded latency as well as support for many-to-many multicast communications.

A number of multicast protocols have been developed over several years to support group communications. While these protocols offer many-to-many services, typical use have focused on supporting applications that require only one-to-many data distribution. Obvious examples include streaming audio and video. Even these early protocols have had limitations in support of more demanding types of applications [Brau93].

RT-DVS applications use visual space management in real-time distributed simulation and supporting communications systems and are evolving to Web based services with XML tagged object characteristics. The performance provided by underlying networks represent an important constraint in deployment of XMSF [Brut02].

The XOM overlay network operational view (OV-1) is presented in Figure 1. The XOM is an overlay multicast system employing protocols designed to support efficient, robust multicast transmissions over existing network protocols such as UDP/IP.

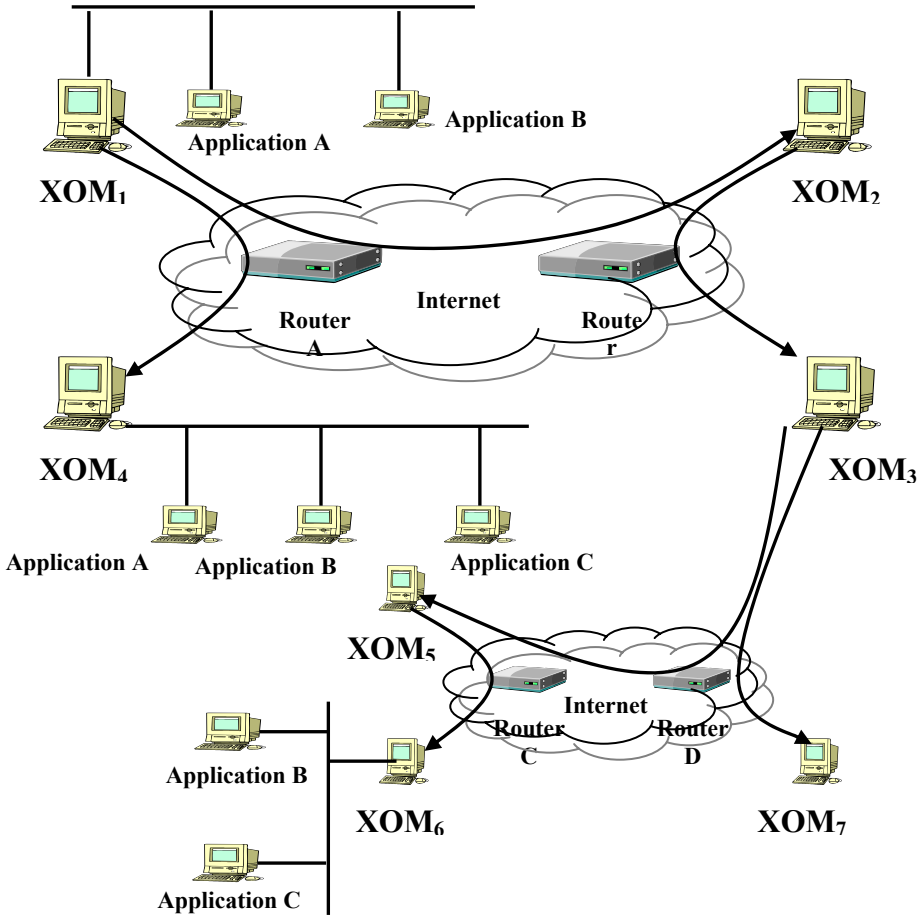


Figure 1: High-Level Operational XOM Overlay Concept (OV-1)

Deploying RT-DVS across many organizations requires robust multicast networking services that are invisible to end users. The proposed approach to XOM recognizes that the underlying networks may have a wide range of network capacities servicing this broad range of users that includes everything from low bandwidth wireless media to modern broadband networks operating at gigabit speeds. The approach also recognizes that, as RT-DVS applications move toward advanced technologies such as XML-oriented Web services as well as agent-based distributed simulations [Wang03], there will also be

a growing need for advanced networking services that are not likely to be available in open networks such as the Internet [Moen03].

The XOM approach must provide the RT-DVS real-time response and predictable network services in order for the end simulation systems to interact within specific delay bounds. Simulations deployed across wide areas nevertheless require low latency, including stringent jitter requirements and high bandwidth demands. Users also desire simplicity in the sense that there should be very little configuration required to allow deployment and establishment of service.

Figure 2 provides an Operational node Connectivity Description (OV-2) that depicts the nodes, activities and information exchanges involved in overlay multicasting. The filled rounded boxes represent nodes. The bullets in the rectangles between two nodes are information exchanges. The bullets next to the node are activities preformed by the node. Any number of nodes could participate in a community of interest overlay where each node would perform the same function as the other organized to provide a multicast service to the application layer.

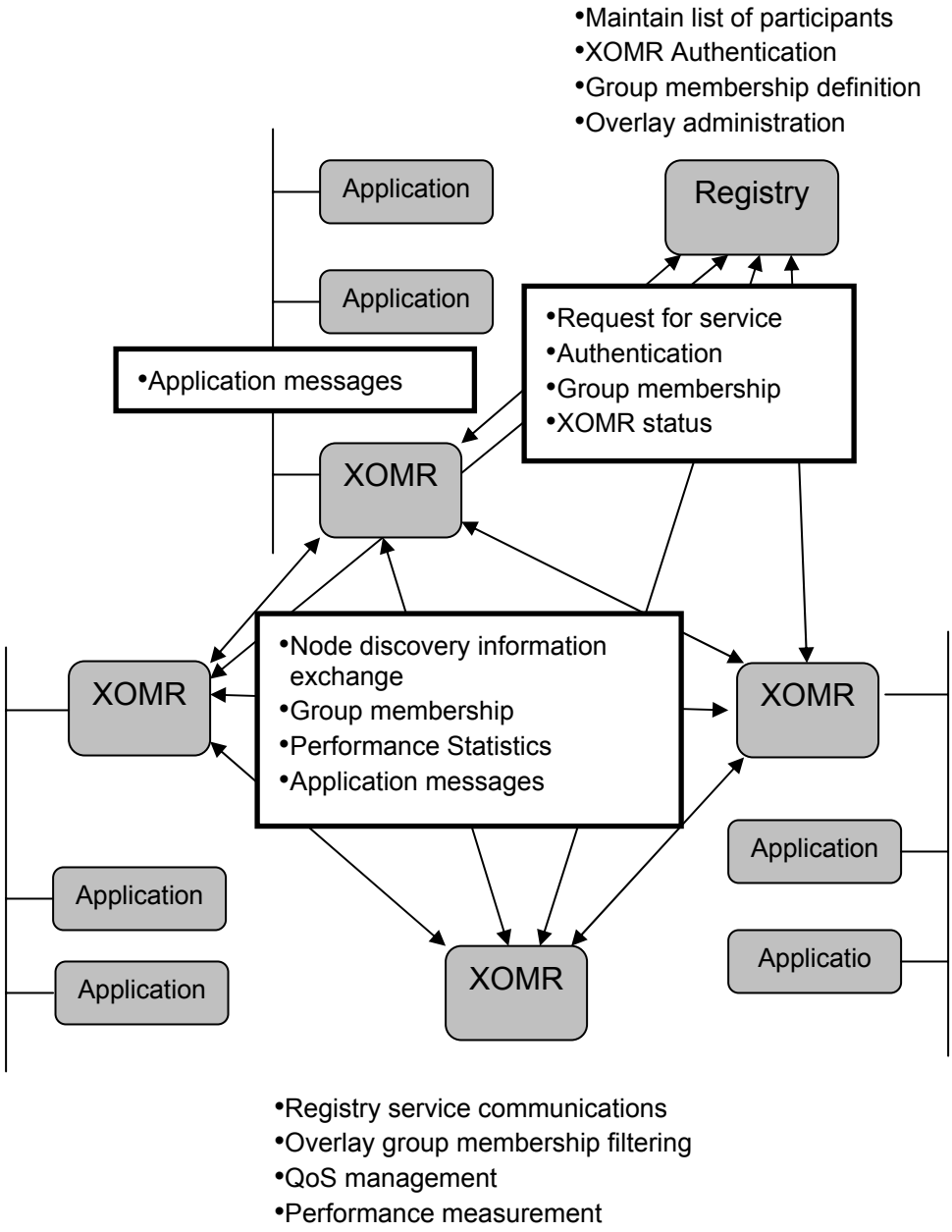


Figure 2: XOM Operational Node Connectivity Description (OV-2)

The following presents a summary of operational description for the XOM in four categories: Group/overlay membership management, QoS, Path management, and Security.

A. Group/Overlay Membership Management

This category requires the XOM to:

- Perform three basic functions in group management: address management, service registration, and group membership maintenance.
- Provide registration services that identify the state of all XOMRs.
- Need to establish and manage membership in a multicast group which implies assigning multicast group addressing scheme for the overlay. All multicast traffic is then delivered to this address(s). This implies that all members of the group must be listening for traffic with this address. In order to maintain compatibility with existing IP multicast, the XOM allows for use of either IGMPv2 or IGMPv3 locally to manage group membership. The organizational community can choose which to use, but must be consistent across the community. Using IGMPv3 allows implementation of source specific multicast where a host joins specifically to a sender and group pair. This capability allows some level of protection to the host from receiving messages that it did not specifically request to receive.
- XOM doesn't provide an inherent address management scheme, so an outside authority (supported by the registry service in XOM) is required to provide the

address of the XOM host. Inherent to XOM this approach is a requirement for an address assignment authority to support local served hosts and provide a service to map multicast requests to an overlay IP path.

- There should be no explicit set-up processing between the sender and the receivers beyond normal IP multicast group joins, prior to the establishment of group communications. An out-of-band mechanism is required to pass the multicast group (IP) address to the associated receivers. The receivers' XOMR will have established support for the address prior to transmission in order to receive the data.
- To add a new user to an existing group, the new receiver must first communicate directly with the supporting XOMR using a mechanism to join a group and exchange relevant information such as the group address. The XOMR adds the new receiver, with the basic connection set-up processing invoked as before, with the new connection completed only if there is sufficient capacity to process the user.
- XOM group membership can be closed by either the sender or the receiver. When the last receiver along a path has been removed, any resources allocated over that path are released. When all receivers have been removed, the sender is informed and has the option of either adding a new receiver or tearing down the group.
- Connection set-up involves negotiation of the path capacity (access capacity) and latency parameters between the sender's XOMR, intermediate XOMRs,

and receiver XOMRs. If the requested resources cannot be made available, the sender is given the option of either accepting what is available or canceling the connection request.

B. Quality of Service

- Diversity and adaptability also must be accommodated by trading quality of service (reliability. latency, and possibly jitter) with the capacity of the access link. Multicast support for quality trades can be realized either through the use of different multicast groups, and/or with prioritization of access capacity in the overlay. Reliability for multi-class traffic can be accomplished through use of a protocol such as SRMP, on top of the multicast overlay or queuing on the send side based on class of traffic.
- The XOM does not provide a flow control mechanism in the context that might be used for bulk data or file transfer. Some higher layer protocol is expected to provide a flow control mechanism to regulate the quantity of data placed on the network based on feedback from the XOM for bulk transfer applications.
- The XOM provides rate control for access to the underlying network service. The service is necessary to allocate available path resources and capacity in a way that maintains the minimum negotiated QoS for the relay agent. Two classes of service are supported: priority and best effort. Using two classes provides a mechanism for application layer to designate priority under

congested or constrained conditions. The rate control mechanism also must provide feedback to the higher layer protocol for application layer flow control. The objective is to provide rate control from the global network perspective based on the network resources used on a per flow or group basis.

- The XOM specification allows the user to determine whether multicast transfers are unreliable or reliable, where reliable transfers are defined to provide a "high-probability of success of delivery to all receivers." SRMP can provide the mechanism to manage this capability for limited amounts of reliable traffic. SRMP, as a transport protocol, runs in the application host.
- The XOM, as an overlay, provides levels of guarantees end-to-end for capacity and latency subject to availability in the underlying network. The guarantees results from implementation of rate control where the XOM dynamically manages the path, ensuring that the available capacity is managed at optimum for the overlay. The enforcement policy ensures that the same path is followed for all transmissions and prohibits new connections over the network unless there is sufficient capacity to accommodate the expected traffic. This is accomplished by maintaining the statistical state of all connections in the XOMR.
- The XOM must acknowledge and be able to respond to the introduction of priority messages above already allocated capacity. The approach is implementation of a conservative statistical approach to capacity allocation

where bursts of priority traffic are allowed within the limits of the current negotiated QoS [Simo03].

C. Path Management

- The XOM protocol suite requires routing support for four functions: path setup, path teardown, packet forwarding, and prioritized packet loss due to congestion.
- The routing tables must maintain both the multicast group address and the forwarding path on each outbound interface in order to make appropriate routing decisions.
- XOMRs receive path setup requests as required when new members join a multicast group. This setup request specifies the incoming and outgoing interfaces, the group address, and the QOS associated with the request. When the message is received, XOMR establishes a path between the server and the receiver, and subsequently updates the multicast group state table and associated port information. Alternatively, the service can be aggregated paths, and not provide sender based trees.
- Path teardown requests also are propagated through the XOMs when group membership changes or QoS changes no longer require data to be sent over a given route. These are used to inform XOMRs regarding both deletions of QoS for a given path and deletions of entire paths. The purpose of the message is to explicitly remove routing table entries in order to minimize the

time required to stop forwarding multicast data across networks once the path is no longer required.

- Interface processes perform send and receive functions between XOMs across the external network and with application hosts on the attached subnetworks (LANS).
- The XOM provides a connectionless service which implies messages maybe sent without permission; hence buffer management/overflow are required at the receive side application layer.
- The XOM provides for two levels of priority traffic: Class B, no priority and Class A, priority.
- Local control: relies on the existence of independent, end-to-end algorithms that can "sense" and react to the distributed, local actions.
- Provide for resource management by periodically gathering and updating information about the service/network.

D. XOM Security

Multicast communications introduce new security challenges compared to unicast communication. RT-DVS multicast applications need source data authentication in order to guarantee that a received message had originated from an authorized source and was not manipulated during the transmission. There are a number of solutions available for normal IP unicast communications. For example, a pairwise security association between one sender and one receiver can provide data origin authentication using symmetric-key

cryptography. In groups, however, a key is required to be shared among more than two members. In this case, a symmetric-key approach does not guarantee data origin authentication. Since multicast implies group associations instead of pairwise, it becomes possible for any member of the group to alter the message. It is therefore important that security services design decisions for the XOM be an upfront design decision. It is important to have security built in, not something added after the fact, includes consideration for protecting the overlay, its members and for providing information exchange protection.

If a systems approach is applied, then it is possible to use the concept of signatures to detect and enable legitimate requests, denying all other traffic. The key security services necessary are:

- Authentication—two processes exchange messages until each process is certain that it is communicating with the other process
- Privacy—each of the processes uses its security key to encrypt any data message before sending it to the other process
- Integrity—before sending a message, the sending process uses its security key to compute an integrity check for the message and attaches it to the message. This allows receiving process to prove the message arrived without modification
- Non-repudiation—sending process computes digital signature to prove that the message is from the sender
- Authorization—check for authorization to use a requested resource/process

Because security capabilities are expensive in terms of both development and operating resources, we specify a minimum working level of security for XOM. First, the minimum requires providing a central authentication of senders via the registry services, a "third party" provider, using existing Internet/Web security protocols to provide a secure channel to distribute a shared secret. These services then allow for membership access control at the subnet level via membership authentication and verification in the context of a specific multicast group. This same approach provides protection of the multicast distribution tree, i.e. the routing protocol that manages and controls the tree. It also enables sender based filtering at the XOMR so that denial of service attacks are minimized to the application.

The second minimal requirement is to be able to provide for a secure channel for application information exchange if an application should desire information protection. This service allows for a sender to have some guarantee of integrity in the message transmission. The Internet protocol approach for this is through encryption normally implemented within a TCP connection using IPsec at the application level or using a gateway to provide encryption at the IP layer.

The recommended approach for XOM is to allow the application to choose this service which implies use and implementation of TCP tunnels. Today there is no known mechanism to provide encryption of UDP for use in a multicast overlay. However, information encryption could be implemented to protect information in a TCP tunnel as a service with the XOMR. This approach requires experimentation and evaluation through

prototyping so that performance can be measured and included in overall XOMR performance expectations.

IV XOM System Functional Description

The system functional description (SV-4) for the XOM is presented in Figure 3 with system function definition following in Table 1. Table 2 presents the data element description for configuration of an XOMR to operate as a node in an overlay network. The XOMR is designed so that each that module can be optimization and alternative strategies for each can be prototyped for evaluation. The approach for the XOM overlay is to use UDP as the underlying network protocol and offer services to the application layer for two different classes of services: Class B-no priority and Class A-priority traffic.

The proposed XOM employs a priority queuing strategy to give priority to class A traffic and mark class B traffic for discard eligible in the event of network congestion. This approach is consistent with our previous efforts in development of multicast protocols such as the SRMP.

Since this approach does not provide error control, any form of desired error control must be added to the client application. The design assumption is that packets are relatively small (<200 bytes) and the underlying network is able to deliver packet guarantees greater than 99% and has reasonable routing path stability on the order of minutes. Reliable transport can also be provided using higher layer protocols such as SRMP, shown in Figure 3 as an example interface, where a more desirable reliability is sought but not available to the client application. Alternatively, the application can

employ measures such as sequence numbers with timeout and retransmission to handle discarded datagrams and sequence numbers so that clients can decide that the datagram is old and a more recent datagram is available or a re-transmission request can be made.

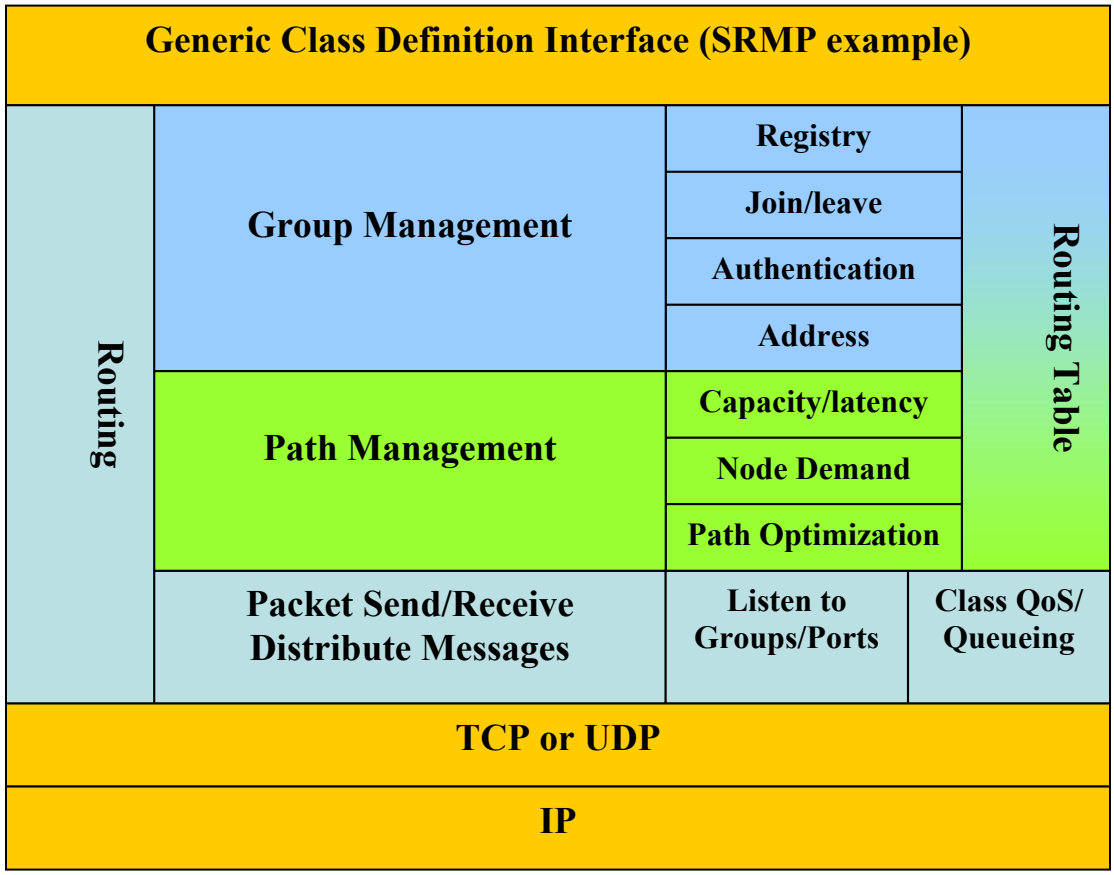


Figure 3: XOM System Functional Description (SV-4)

Table 1: Function Definition/Operational Activity Matrix

Function	Definition	Operational Activity	Data Exchange
Generic Class Definition Interface	Interface between XOMR and application	Provide interface JAVA socket to pass message data to/from XOMR	• Send message to Group request • Receive message
Routing	Message forwarding/receiving to/from nearest neighbor XOMR	Computes routing information and forwards messages to nearest specified neighbor XOMR	Message forwarding specified by next hop address in routing table
Group Management	Provide services for managing multicast group membership	Process group membership requests and maintain group status information	Group address
Registry	Central control for community of interest for overlay	Receive/approve request for overlay membership, announce group membership, maintain and distribute table of active XOMRs	• Registration service request • Registered XOMR status information
Join/leave	Group membership process for application to join and leave desired multicast groups	Process algorithm for managing group membership in the overlay	JAVA Service request
Authentication	User/host authorization for joining a particular XOMR/multicast group	Provide a secure environment for host/application to prove to the overlay that it has permission to send/receive	XOMR Host/application identification using Authentication Authorization Accounting (AAA) or similar mechanism services of Registry host.
Address	IP address of host requesting service	Assign IP address	Public routable IP address

Function	Definition	Operational Activity	Data Exchange
Path Management	Management of optimal overlay multicast routing distribution tree	Distribute source based routing information to all nodes in the overlay	Routing table
Capacity/latency	Access link capacity and end-to-end path transmission delay	Measure access link capacity and end-to-end path delay from to destination	• Bits per second • Milliseconds
Node Demand	Degree or measure of the number of neighbor XOMRs that an XOMR is able to replicate messages to.	Manage ability of host to replicate messages relative to desired performance (message) loss at the node	Number of message replications required for each received message
Path optimization	Routing algorithm for construction of optimum source based distribution tree	Calculate optimal tree for the overlay at each node	Next hop IP address in routing table
Routing Table	Table of rounding information used by XOMR in determining next hop forwarding address	Store routing information calculated by path optimization algorithm	Next hop IP address
Packet Send/Receive	JAVA socket interface between XOMR and lower layer protocol	Send/receive messages to/from lower layer protocol	Messages with routing information header including group membership
Listen to Groups/Ports	JAVA service interrupt for message flow	Threaded object that listens on a UDP or TCP port for traffic. When echo or data traffic comes in it parses the header and hands it off to the appropriate object for processing	System interrupt
Class QoS/Queueing	JAVA buffer for management of send/receive messages	Manage buffer overflow based on class of service specified by user.	Buffer size

Function	Definition	Operational Activity	Data Exchange
TCP or UDP	Internet transport protocol specifying type of service and flow control	Transform application message to meet Internet protocol specification for IP layer using UDP or TCP based tunnels.	• UDP protocol format for best effort service • TCP tunnels for streaming data and IP based encryption
IP	Internet Protocol	Message routing across Internet (open network)	IP packets specified by IETF format

Table 2: Data Element Description for XOMR (SV-4)

Data Element	Description
registryAddress	InetAddress of registry, 0 if none
numberOfMulticastGroups	count of groups/ports we will support
numberOfPortsPerGroup	count of ports each group will support (non-overlapping)
lowestMCAddress	first group address to multicast from the subnet, dotted decimal notation (other addresses follow in sequence)
lowest port	first UDP port to multicast (each address will get one port in sequence)
routingUpdateInterval (optional)	time in ms between routing updates (default 10 s)

Data Element	Description
thisSubnetMaskBits (optional)	number of bits used for routing in subnet address (default 24)
useTCP (optional)	0 for UDP tunnels, 1 for TCP tunnels (default 0)
partnerHostAddress (optional in future, when Registry becomes available)	zero to MAX_PARTNERS IP addresses, in dotted decimal format, to be used as partners without checking the registry

The core of the XOM is provided in the Group and Path management functions as the result of these activities provide the information for the routing table to be used in making packet forwarding (routing) decisions. Three approaches should be considered for managing the multicast path overlay and associated groups:

1. The XOMs could provide a service that is independent of group management and essentially provide a path(s) overlay optimized across an open network to other registered XOMs. Under this model, the path overlay looks like a closed network with all group communications provided as single multicast network similar to Protocol Independent Multicast – Sparse Mode (PIM-SM) specification [Estr98]. Each user application of the XOM, then listens for desired group identification communications broadcasted on the local area network hosting the XOM and discards/ignores unintended traffic.
2. The XOM could provide a service that recognizes group membership dynamics (registered XOMs that host users/applications identified by group) and provide an

overlay path optimized for each group. This approach generally is referred to as source-based tree multicast [FeiA01].This implies management and optimization of multiple paths, essentially a path structure for each registered multicast group. The current XOMR prototype uses this approach.

3. The XOMR can provide a service that aggregates group traffic across optimized paths between XOMRs as presented in Figure 4. These optimized paths are essentially shared trees and can be optimized for capacity and delay to support aggregated group traffic [CuiJ04]. (See Appendix B for description of concept for building overlay trees.) This approach also is similar to aggregated group multicast over MPLS [CuiJ04] and with added features for group management. The current XOMR prototype performs aggregation.

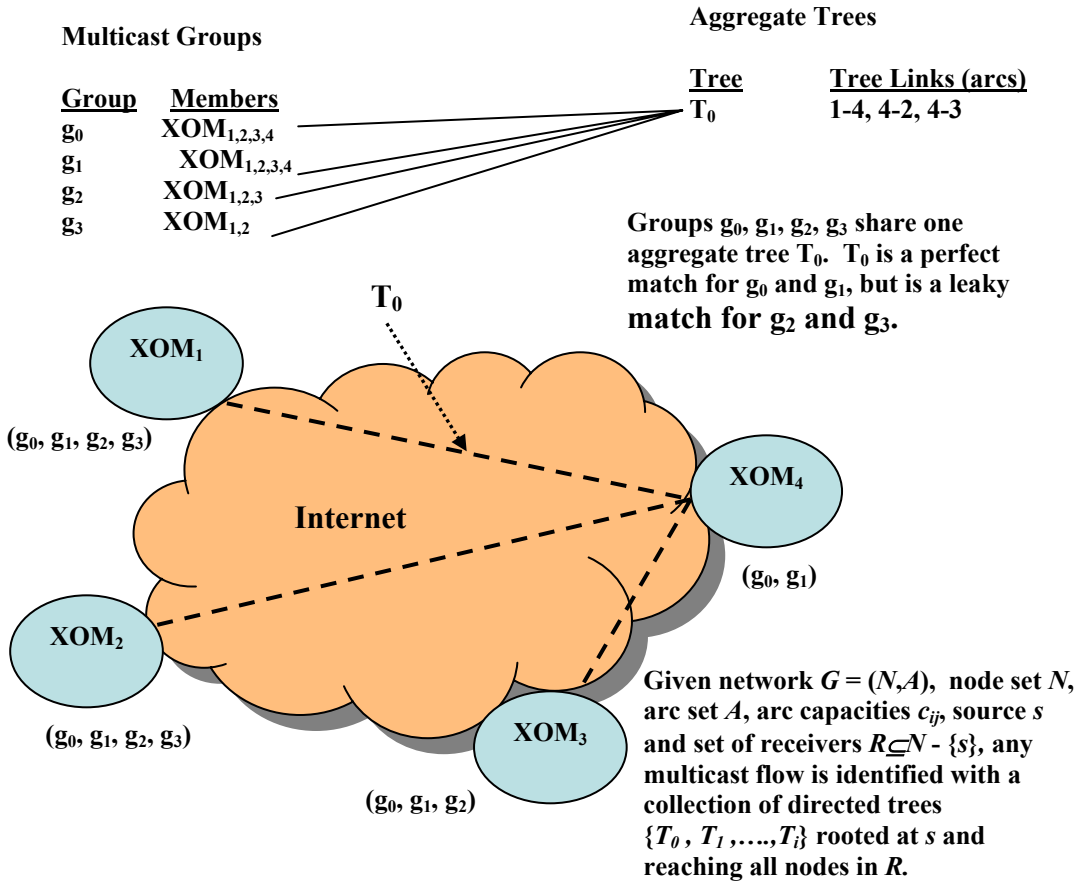


Figure 4: Group Aggregation Overlay

In all cases, once an XOMR and associated address is established, receivers will issue a request to join existing groups using a unique connection identifier that is pre-assigned by the registry. Using this approach, the RT-DVS application is able to control or specify which sources of information are of interest. This is important because we expect that many disparate RT-DVS applications could conceivably be using the same local supporting XOMR. This helps protect an individual RT-DVS application from receipt of unspecified or undesired information flows and also aids in minimizing overall network

traffic load. This will require that we specify how the XOM identifier is allocated and how the receivers learn its value the external registry service and supporting protocol.

At the local level, the XOM manages the receivers' interests in receipt of group messages. This feature allows for a host to report to the local XOMR interest in receiving packets only from specific source addresses and therefore aids in the overall management of group membership and optimization of traffic loads on the network. This approach also potentially adds a level of security to the RT-DVS application as the application is able to discard or ignore messages from unauthorized sources.

We allow for the join request from a RT-DVS application to specify whether the receiver wishes to be a producer of information as well as a receiver, whether the connection should be able to provide the two classes of service (no priority or priority), whether the receiver is able to accept multiple senders of information, the minimum throughput desired, and the maximum data message size. This request information is presented to the supporting XOM and used in the group join process to support negotiation of the path capacity and latency parameters among the sender XOM, intermediate XOMs, and receiver XOMs. If the needed resources to support the request cannot be made available, the sender is given the option of either accepting what is available or canceling the connection request.

An application request for terminating membership in a group is coordinated through the supporting XOM. XOM connections can be closed by either the sender or the receiver. When the last receiver along a path has been removed, the resources allocated

over that path are released. When all receivers have been removed, the sender is informed and has the option of either adding a new receiver or tearing down the group.

We have not included in the specification what action the local XOM should take when the application group is reduced to a single member, but a logical action would be to stop transmission if there are no active receivers and announce this to the registry service. It is also possible that the XOMRs will discover this by communicating with each other, rather than via the registry. In this case, the routing protocol will prune a dead-end XOMR from the routing tree but keep an XOMR if it is making the tree more efficient by providing a branching point.

Our group membership approach assumes that a group definition is based on a specific application running behind an XOM on the local area network. Multiple instances of an application are supported behind each XOM, each of which may have different group membership characteristics to include membership in multiple groups. It is also feasible for a RT-DVS application to have membership in more than one group. We present an example in Figure 5. Notice that application B has membership in group 1 and in group 2. In order to maintain efficiencies in packet transmission, we form a new group 3 that is the union of the two groups. We also imply no explicit set-up processing between the sender and the receivers prior to the establishment of group communications. The XOM mechanism is required to pass the multicast group (IP/group tag) address to the XOMRs of the associated receivers. The receivers' XOM must have established support for the address prior to transmission in order to receive the data.

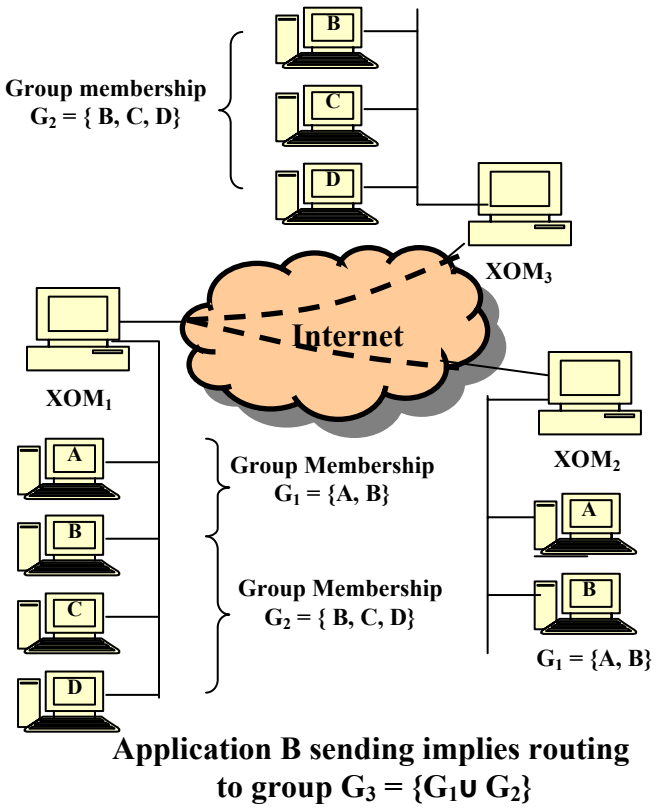


Figure 5: Group Membership

Allocation of a specific XOM IP address, or network service access point requires the use of an outside addressing/registry authority to establish an XOM host. The operational concept of the registry is presented in figure 6 with the data structures presented in figures 7 and 8. The registry is responsible for keeping track of relay participants in an overlay network. On startup, a relay contacts the registry via SOAP-RPC. The registry returns a list of all the other participants in the overlay network, and notifies the other participants that a new relay has joined using the data structures presented in figures 7 and 8. This provides a level of security by establishing an authority that authorizes a

source to be a sender which can be used by networked XOMs receivers for recognition of authorized senders in the network. The registry also will maintain the public routable IP address of all active XOMRs to be used by the XOMRs to establish efficient overlay multicast paths between XOMs. The registry also will maintain multicast group membership information. Once an XOMR host is established, internal protocol mechanisms provide for path optimization among XOM hosts and manage multicast group membership at the local XOM.

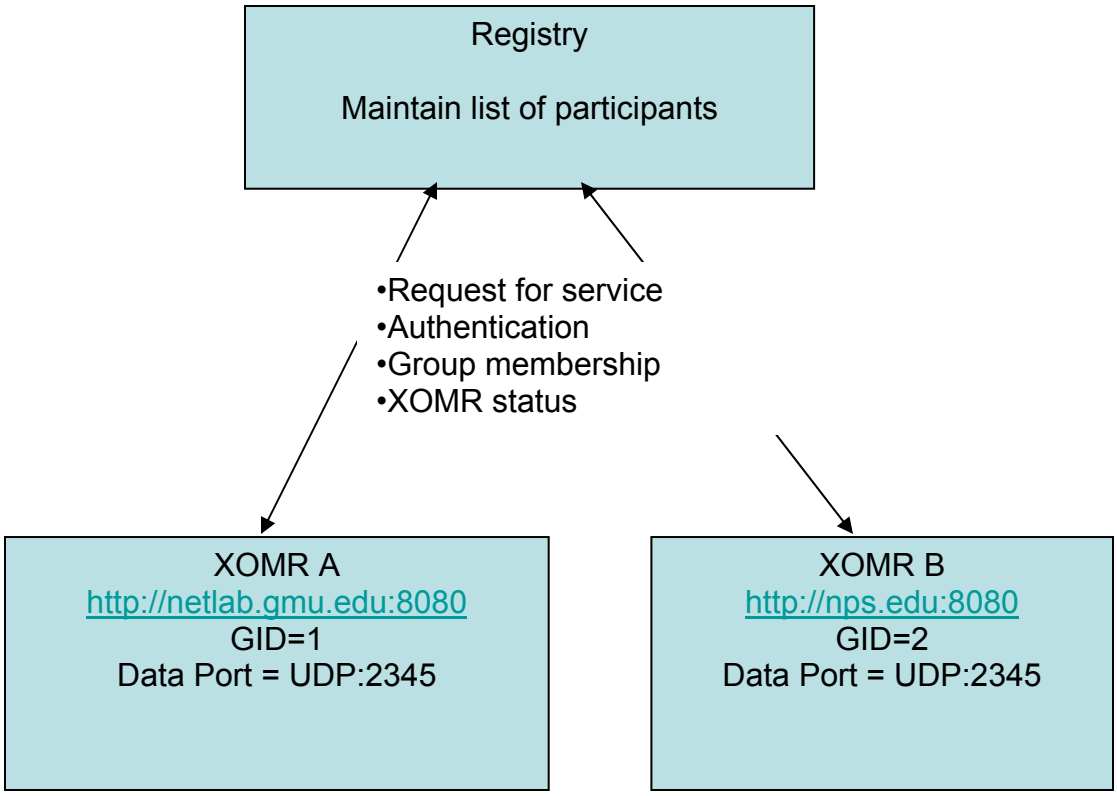


Figure 6: Registry Notional System Functional Description (SV-4)

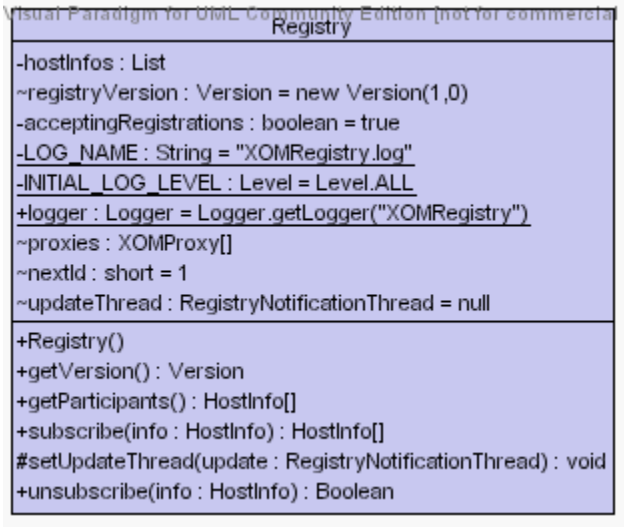


Figure 7: Registry UML Data Description (SV-4)

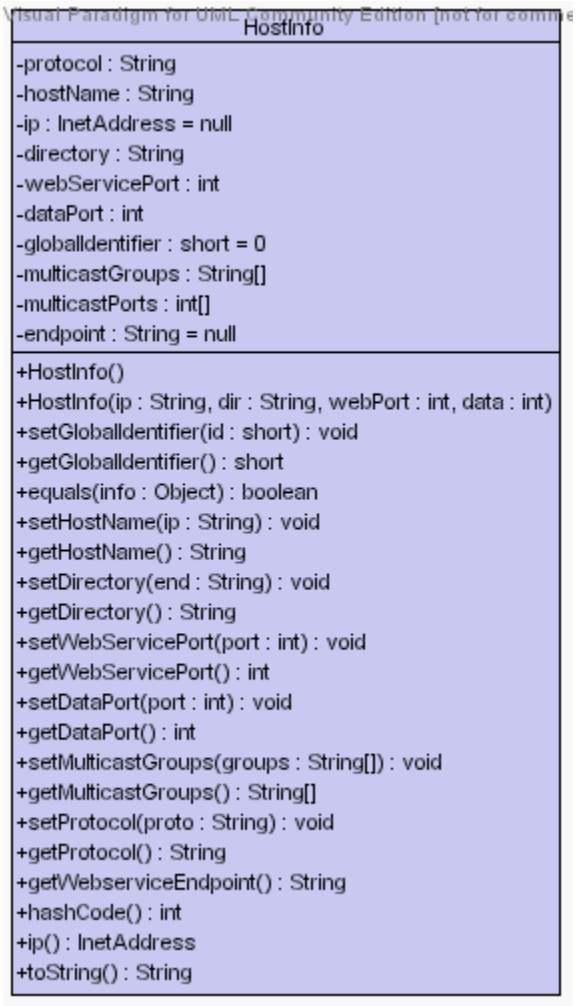


Figure 8: XMOR Host to Registry UML Data Description (SV-4)

V XOMR Routing Protocol

A. Overview

The routing protocol defines the method used by the XOMRs to exchange reachability information. Its key elements are the use of a central registry and the local XOMR that provides overlay network multicast services to supported RT-DVS

applications. The registry maintains a list of all XOMRs in the network and registered RT-DVS application users with their requested group membership, but is not required to maintain the topology of the overlay. The only requirements are that the registry responds/replies to all requests from an XOMR and any valid XOMR can send messages at least to its neighbors in the overlay network.

XOM relies on three steps to build the overlay. The first step is that a joining XOMR must send a request to the registry access to the overlay. Second, the XOMR must discover neighbor XOMRs that are potential candidates for the joining XOMR to establish a network connection with, essentially building and becoming part of an overlay mesh. The third step is for the joining XOMR to establish the services necessary for group management and exchange this information with networked XOMRs, calculate (and update routing table) optimum paths (shortest path tree) for group multicast routing from source to group destinations, and propagate that routing to all other XOMRs.

There are two mechanisms that contribute to global service guarantees. The first is to limit the out degree of an XOM host using Bollobas' definition of the degree of a vertex [Boll01]. That is, we do not allow the construction of more than *n* connections to other XOMRs. This serves to limit the processing demands and network access capacity of individual XOMRs in the overlay. The second is to threshold the end-to-end overall path delay from a sending XOMR to a destination XOMR and offer only best effort above this threshold to joining XOMRs that do not successfully find an existing XOMR node that is adequate to maintain end-end-to-end path delay thresholds (e.g. 110msec threshold in networking latency, based on the fact that 150 msec is representative of human response

times for RT-DVS. This can also be accomplished by establishing a network diameter threshold based on Bollobas' definition of network diameter [Boll01].).

We provide congestion control by providing two levels of service to the RT-DVS application. Class B packets have no priority and Class A packets have priority. We apply weighted fair queuing giving priority to Class A packets in the send queue of the XOMR and discard Class B packets randomly during congestion.

Since the XOMR is hosted on the LAN that connects to the supported RT-DVS application, we use IGMPv3 [Cain02] for group management at the local level.

B. Network Protocol

1. **Central Registry.** A central registry provides a service to register the presence of a XOMR and the participation of an RT-DVS application. The central registry maintains a cache of all nodes participating in the overlay.
    a. The registry maintains the IP address of the XOMRs and authenticates and audits continued participation in the overlay.
    b. The central registry is reachable by all XOMR nodes at all times.
    c. In the request for join by a new XOMR, the exchange messages will provide for measurement of round trip delay time (RTT) between the registry and the joining XOMR. The registry will maintain this time and periodically update the information.
    d. On request of a new XOMR to join the network, the registry will provide the joining XOMR the addresses of existing XOMRs. The XOMR then

randomly polls the existing set of XOMRs measuring the RTT. The responses with the shortest RTT represent the initial best candidates as the nearest neighbors in the overlay. The XOMR continuously performs random polling of known member XOMRs of the overlay, always looking to optimize the selection of best neighbors.

e. The registry provides for registration of the RT-DVS application, authenticates and authorizes relationship to a serving XOMR as a legitimate source/sender, assigns a node ID, and maintains group membership participation based on RT-DVS application request.

f. The registry group management service provides for creation of a multicast group and assigns a group ID using the IP multicast address as the group ID. This information is then made available for source and receiver RT-DVS applications. A joining host source/receiver application can then use the information locally to indicate to the XOMR a desire to join a group by providing the group ID to the XOMR (XOMRs will use IGMP for this function locally) [Bhat03].

2. **XOM Overlay Construction.** The XOM constructs an overlay using a decentralized algorithm by searching for potential target existing network XOMRs to become the child of in the overlay (nearest neighbor). It uses measures of RTT to candidate neighbors to make decisions on joining a parent already in the network.

a. An XOMR ready to join a multicast overlay sends a request to the registry indicating desire to.
b. The registry authenticates the XOMR against a previously established authorization in the registry and responds to the joining XOMR with a list of all XOMRs in the existing network supporting the desired group membership.
c. The XOMR sends echo requests to *N* candidate XOMR partners resulting in RTT replies.
d. If the candidate XOMR existing connections are less than *n*, the parent XOMR responds with a message indicating availability or else ignores the message.
e. The joining XOMR uses the message exchange to measure RTT to the candidate(s) partner XOMR(s). With the RTT information, the joining XOMR selects the best candidate as the primary connection and selects the second best as an alternate path and responds to these potential parents with acknowledgement of the selected connection and ignores all other responses. The primary partner and the alternate send acknowledgements to the joining XOMR to complete the network join process.
f. The partner XOMR updates the routing table to reflect the new neighbors with measured latency and propagates this information to all its neighbors in the overlay. The result is a connected graph.

g. The XOMR uses the designated primary path unless it is not available in which case it uses the alternate neighbor connection.

h. The XOMR maintains a routing table which maps a node ID to the node IP address and next hop, e.g. neighbor along the path to the distant node.

i. Periodically each XOMR sends heart beat probe messages to its neighbors to determine if the neighbor is still connected and if necessary, initiates the node join process in the case of disconnected nodes by sending a request to the registry, or by probing known XOMRs in the network.

j. Periodically, each XOMR sends random discover messages to other known XOMRs to discover if a better neighbor (link cost) is available and makes decisions on alternate path choices over using the current path. To support this, the registry must periodically update the list of known XOMRs in the overlay.

3. **Link Delay Measurement.** The XOMR develops the link delay data between itself and its neighbors by measuring RTT. This information is shared with the XOMR neighbors and propagated across the overlay.

    a. The XOMR collects link delay information shared and periodically updated from neighbor XOMR.

    b. The link information is propagated across the overlay so that each XOMR will have knowledge of primary and alternate link delay information between XOMR nodes in the connected graph. It is not necessary to know each link weight in the graph, if all the nodes in the graph have distinct

identities. However, distributed minimum spanning trees are constructed with fewer messages if this knowledge exists at the source node [Gall83].

4. **Multicast Tree construction.** The overlay constructed by the distributed algorithm is a connected undirected graph with ***N*** nodes and ***E*** edges, with a finite weight assigned to each edge. A distributed algorithm is used at each node to determine the minimum-weight spanning tree (MST)
    a. We desire the algorithm to optimize overall performance across the entire overlay. This means optimizing certain objective functions to improve overall quality of message delivery. This quality is typically measured in terms of latency and message loss. The algorithms solve a shortest path problem potentially with many constraints to build a tree connecting the source(s) and destination group members so that minimal message flow occurs in the distribution of the message as well as maintaining optimum or minimum latency form source to each destination. The optimal case is that only a single copy of the message flows on any link in the overlay, meets application latency requirements, and offers some level of reliability. Typical resources that are allocated or that must be optimized are link capacity, host processing capacity, number of links or diameter of the tree, and the degree of a node in the path overlay. In addition, the algorithm must lend itself to supporting dynamic overlays where many multicast group members join and leave in real-time. It is also important to consider the scalability of the algorithm which is made considerably more

difficult because end systems typically have limited topological information in which to construct good overlay paths. All these factors are considered in the optimization. A number of algorithms have been considered, for example the constrained Steiner tree [Komp92] and distributed delay bond algorithms [JiaX98]. If we run the optimization algorithm at each node, we obtain aggregate multicast paths from each XOMR as a sender to all other XOMRs.

b. The addition of a new node requires the new node initially join as a child node as described in the XOMR overlay construction procedure above. The join information is propagated to neighbor nodes and the new node runs the distributed shortest path algorithm and sends routing table updates to the XOM overlay.

5. **Node Departure.** There are two cases for node departure. A network event may result in the XOM not being available in the overlay, in which case, the routing algorithm must be able to repair the overlay using the alternate link to nearest neighbor. In the second case, an XOMR may no longer have supported group members and at some point may desire to log off the overlay. This case represents a soft leave and allows for message exchange to effect new path construction without the disruption of service.

VI Summary

Widely deploying RT-DVS across many organizations with large numbers of applications requires robust multicast networking services that are transparent to end users. This report has described a top-level architecture for the XOM that recognizes that underlying networks may have a wide range of network capacities and capabilities and do not necessarily offer a multicast service. The proposed architecture provides a multicast service to higher layer applications that require this capability across open networks. The approach includes consideration for reliability by providing two classes of services on top of existing UPD/TCP/IP protocols.

The overlay multicast middleware is defined as the XOMR where relay implies forwarding or routing of messages to designated destinations from authorized sources. A relay host resides on every subnet that is host to applications that desire multicast services between them. The XOMR host provides all controls and management services necessary to provide the multicast service. The architecture includes provisions for a registry service that enables security features so that only known XOMRs have access.

APPENDIX A OVERLAY MULTICAST RESEARCH AND ANALYSIS

SECTION I INTRODUCTION

I Background

Distributed virtual simulations operating across a network in human time generate large amounts of message traffic between the computers hosting the simulation applications. This requires many-to-many communications in a dynamic group environment where N computers in the group scales as O(N) message transmissions from each member or $O(N^2)$ total message transmissions in the group [Pull99]. In addition, this simulation environment may not necessarily be homogenous, e.g. each simulator is likely to be different but they dynamically share common simulation objects over time. The result is that simulation objects may have membership in multiple groups with each group's membership changing at different rates.

RFC2502 [Pull99b] describes key networking requirements for distributed simulation that result from the interaction of humans participating in these simulations in real-time. Networks supporting these must distribute large amounts of data within the bounds of the human interaction time which leads to the need for specific delay bounds. The environment requires many-to-many distributions of data where many senders are sending to many receivers simultaneously. This environment can be described as a multiparty collaborative environment supporting multimedia applications. The underlying

networking environment needs to support a large number of participants dynamically joining and leaving the communicating groups across the myriad of public and private networks that make up the Internet. Because each of these networks is independently managed, the Real-Time Distributed Virtual Simulation (RT-DVS) applications cannot solely rely on the Internet to deliver the necessary Quality of Service (QoS) even though QoS mechanisms are beginning to become available. As a result, networking real-time simulators together has seen limited deployment, and then only in specialized local area networks or on private networks dedicated to the simulation environment.

Similar limitations apply to running any application that requires group communications across the Internet. In response, a number of researchers have proposed an *end system* approach for supporting multicast across the Internet where all multicast services are provided by the end system rather than by lower layer network services [ChuY01] (Appendix A provides a detailed explanation of end system multicast.). Because the overlay provides a place to manage QoS, it is likely that a RT-DVS application running an end system multicast protocol would be better able to take advantage of underlying QoS protocols across the Internet. Since this approach is independent of underlying network improvements, the overlay approach would automatically inherit any improvements in QoS of the Internet.

This approach may also help expand the user base since the concept could extend to individual personal computers (PC) acting as “network nodes” in the end system multicast schema. The introduction of low cost, very fast processor PC’s as well as the growing presence of lower cost broadband access has the potential to make this a reality.

After many years, IP multicast is still not widely deployed over the Internet. In fact, there are inherent disincentives for public carriers to implement IP multicast in their networks, the largest being a financial incentive. Public carriers invoice based on fixed or burstable amounts of traffic where as multicasting is designed to reduce offered traffic load. While the IETF is exploring one-to-many multicast over the Internet as there is a growing support for this service, it is unlikely that many-to-many multicast over the Internet will ever materialize, as there is no clear business case. There also are other concerns for broad deployment of IP multicast related to reliability, congestion control, network-to-network policy sharing, and scalability [ChuY00].

These concerns have lead to a growing support for end system multicast where all multicast related functionality is implemented in the end system or the host in an overlay structure. In this overlay, all functionality of multicast, including group management and message replication, are performed in the end host. Further, these end systems approaches include optimization of the overlay by adapting to network dynamics and are able to take application performance into consideration [ChuY00].

Studies have been undertaken to view performance of an end system approach for the class of applications supporting audio and video conferencing [ChuY01]. Results of these studies are highly favorable for this approach using one-to-many multicast services. Audio and video conferencing applications have very stringent performance requirements for delay and throughput similar to those required for real-time distributed simulations. Distributed simulations operate with human participants who expect to experience a

virtual presentation in real-time [Pull99], not unlike the requirements of audio or video conferencing.

However, the RT-DVS environment is significantly different from audio and voice conferencing in that jitter is generally managed in the audio/video end system application layer, by buffering for the one-to-many multicasting. In RT-DVS, jitter generally is managed using time stamps in the application.

For RT-DVS end-to-end system performance, there still remains a higher-level requirement to provide a managed multicast service across the many independent Autonomous System (AS) domains the make up the Internet. To achieve this, every AS domain manager must have an incentive to maximize performance across their managed domain. The choices made by each manager must not be hindered by any other AS in order for the end-to-end system application to achieve desired results across the network. Overlay multicasting is a way to overcome this limitation and provide independence from AS domain management. With this independence of AS, an overlay multicast protocol is able to take advantage of the service offered by QoS facilities (e.g. Multiprotocol Label Switching (MPLS) or Differentiated Services (DiffServ)) in the AS without even the necessity of knowing that the AS is using this underlying technology. This approach provides the network independence necessary for a set of real-time distributed simulations to interact effectively within the confines of the network overlay.

II Statement of the Research Problem

The thesis of this research is that it is possible to use an end-system multicast approach to provide the necessary multicast service across an open network to support RT-DVS many-to-many multicast requirements. This includes providing a predictable network performance that meets the stringent requirements for capacity and delay with many-to-many multicast capabilities, subject to adequate performance in the underlying network. Further, this approach provides an attractive alternative to network based multicast as this approach allows the distributed simulation to become independent of network based multicast. The case for this approach includes the historical end-to-end argument [Sava99] that functionality should be (a) pushed to higher layers if possible, (b) unless implementing it at the lower layer can achieve large performance benefit that outweighs the cost of additional complexity at the lower layer.

To verify this thesis the following items were developed:

- A message traffic generator for use in performance evaluation of a prototype protocol used to demonstrate feasibility for the RT-DVS application environment.
- A study and analysis of actual real-time simulation environments to characterize expected message traffic load.
- An analytical model of the multi-flow characteristics to determine the regions of feasibility for performance of the underlying network service required to meet the real-time simulator requirements for distributed human time simulation

A. Value to RT-DVS Environment

The ability to perform many-to-many multicast over an open network is very important to the RT-DVS community and is essential to implementing the Extensible Modeling and Simulation Framework (XMSF) [Brut02]. This framework, recently recognized by the Simulation Interoperability Standards Organization (SISO), has the objective of using XML and web based technologies for expanding the user base of RT-DVS. Implementing end system multicast for real-time distributed simulations allows the continued use of open protocols as implemented across the Internet. RT-DVS is then no longer dependent on consistency of network policy implementation for multicasting across an open network such as the Internet and supports the RT-DVS community's effort to move to web based technologies such as XML.

B. Challenges to the Problem

There are many complexities of the RT-DVS environment that are not readily supported in the current Internet. As indicated earlier, many-to-many multicast is required and at this time and for the foreseeable future, is not available in an open network environment even for a small number of users. In this many-to-many environment, sources can be receivers and vice versa. Both can act as "virtual" nodes or routers in the network. A small number of users can precipitate a very complex multi-path problem.

The RT-DVS also requires that QoS be specified or negotiated for at least capacity and latency, and if possible, jitter, and packet loss in a statistical sense, end-to-end or

between applications. In general terms, QoS can be specified or negotiated within private networks or individual ISP (Internet Service Provider) networks, but not across the open Internet. In these private networks, protocols such as the Resource Reservation Protocol (RSVP) can be employed to provide QoS services. For Internet wide QoS negotiation, no known strategy exists. Achieving certain QoS objectives also implies tradeoffs. For example, reliability and latency work hand in hand as increased reliability implies greater latency. It is impossible to have fully reliable/real-time multicast in the general case because the acknowledgement mechanism used for reliability would overwhelm the sender if many receivers are involved. Reliability is available only in the form of a selectively reliable/real-time or fully reliable/non-real-time capability.

Finally, there is an added dimension of complexity in supporting distributed simulation. Classic one-to-many multicast to support video or audio streaming generally can be supported with minimal spanning tree routing. Current routing protocols use such an approach. Because RT-DVS demands a higher level of performance management from a global optimization perspective, the new problem becomes one of a dynamic extension to the multicast tree problem. Here global means all the participants in a common simulation exercise or experiment operating across an open network. There are two special cases that result in the problem being dynamic for this environment:

- Network events: disruption, congestion (QoS requirements not being met), and/or spare capacity re-allocation optimization may result in the cost of the edges in the current tree to change or the cost of the edges not currently in the tree change as a result of multicast bandwidth re-allocation since the current tree was first

computed. This results in the need for dynamic re-calculation of the tree. While these events are similarly disruptive to streaming audio or video applications, these applications tend to have one-to-many distributions so that tree recalculations are less protocol intensive.

- The RT-DVS environment has a very dynamic multicast group characteristic. Objects of the virtual simulation join and leave many multicast groups frequently as the scenarios of the simulations are executed. This dynamic group characteristic may result in the frequent need to re-calculate the multicast tree to re-allocate network capacity or generate new paths or new flows within paths to support the changes.

C. Problem Solution Considerations

The RT-DVS is dependent on real-time response and predictable behavior in order for the end systems to interact with the physical world within specific delay bounds and present data such as images, audio, and video on a real-time basis. The users are expected to be deployed across the Internet and/or intranets and have low latency including stringent jitter requirements and high network capacity demands. To support this environment, there are a number of factors to be considered in developing the overlay architecture for the RT-DVS.

1. Traffic Characterization

A traffic model with appropriate metrics is required to describe the behavior of the RT-DVS. These metrics must take into account packet error and loss rates, throughput, latency, and path flows. No historical data is available to characterize the traffic generated by RT-DVS applications. Generally, this is complicated by the nature of the application as the traffic load is specific to a simulation scenario. To facilitate studies of traffic load behavior of protocols supporting the RT-DVS, a traffic model has been developed for use in the GMU C3I Center Networking and Simulation laboratory (NETLAB) [Moen01].

2. Demand Characterization

A model to describe resource demand is required for a scalable solution. This includes network capacity assignment at each server and dimensioning/quantifying the load at the servers according to a multi-flow session traffic model. The scalability question to be addressed is the required capacity of the link(s) that connects the server to the Internet, such that acceptable service levels are achieved and maintained. Modeling this requires implementation of an open multi-flow network model where C is the number of flows and λ_c (c = 1,2,3…, C) is the arrival rate of flow c. Based on this model, service demand in terms of capacity must be calculated.

3. Path Characterization

A routing algorithm is required for discovery of valid network paths and associated capacity. The objective is to manage the exhibited capacity experienced by the local host. This approach simplifies path management and allows the local host to make decisions on ability to support demand for available resources.

4. Path Convergence

The approach requires dynamic overlay network optimization, as this is a very robust environment requiring many-to-many multicast with multicast group members joining and leaving the groups real-time. The problem is to dynamically find a set of paths that connects a finite set of points in a metric space with the shortest possible length. The problem is difficult given end systems typically have limited topological information in which to construct good overlay architectures (paths). The problem is made more complex in that the approach could allow insertion of additional nodes to achieve optimization. This problem is more commonly known as the Steiner problem [Floy99, Onoe97]. The general problem can be defined as:

> Given network $G = (N,A)$, node set N, arc set A, arc capacities c_{ij} (expected message delay on an arc), source s and set of receivers $R \subseteq N - \{s\}$, any multicast flow is identified with a collection of directed trees $\{T_1 , T_2 ,\ldots,T_l\}$ rooted at s each representing a one-to-many distribution tree across the network and reaching all nodes in R and associated flows $\{x_1 , x_2 ,\ldots,x_l\}$ with constraints:

$$\sum_{(1\le k\le l)} f(i,j,T_k)*x_k \le c_{ij} \quad \forall\ (i,j) \in A$$

$$f(i,j,T_k) = \begin{cases} 1 \text{ If arc } (i,j) \text{ appears in Tree } T_k \text{ and } 0 \text{ Otherwise} \end{cases}$$

This multicast flow problem involves sending flow from a source to a set of destinations or receivers (one-to-many). A multicast flow is then defined as a collection of flows along directed trees obeying the capacity constraints for each arc in a path.

III Research Approach

The research strategy applied to this problem started with a definition of the concept through literature research and then defined through analysis and modeling, the value of the concept. The research strategy included five steps:

- Collected data and performed analysis of live simulation applications using OPNET modeling data capture and analysis capabilities.
- Developed the network service requirements for the RT-DVS community such that an over-lay, end-to-end approach for QoS based many-to-many multicast can be defined.
- Based on the RT-DVS network service requirements, developed a top-level architecture for an end-to-end managed QoS, many-to-many multicast routing mechanism that will support the stringent requirements of RT-DVS.
- Developed an analytical model to support performance analysis of the basic concepts.

- Validated the analytical model and the approach by performance measurement of a prototype in the laboratory and across the open Internet.

IV Summary and Unique Contributions of this Research

The unique contributions of this research are in the areas of traffic modeling and performance evaluation of a proposed strategy of using multicast overlay as a mechanism to support many-to-many multicast for real-time distributed simulation. The specific contributions include:

- A methodology for modeling the traffic generated by real-time simulators
- An analytical model that characterizes traffic flow in a multicast overlay network
- A characterization of actual message traffic from the study of live simulation environments and
- A top-level proposed architecture for a multicast overlay protocol called XOM to support RT-DVS.

Implementing a multicast strategy that maintains RT-DVS independence of the underlying network architecture that at the same time takes advantage of Internet QoS implementations is a highly desirable strategy for RT-DVS. The outcome of this research provides evidence to support the viability of an End System multicast architecture with QoS management for enabling performance demanding RT-DVS across the Internet.

Section 2 provides a background on a number of other on-going initiatives related to overlay multicast. The initiatives are presented in the context of satisfying a specific application environment in the same fashion that this work is founded. Included in this

section is a discussion of different strategies for overlay multicast to include ideas such building meshes versus trees and using combinations of differing strategies.

Section 3 presents a background on multicast routing algorithms used in many routing protocols. The discussion includes concepts about optimizing certain objective functions in routing algorithms to improve overall quality and efficiency of message delivery. The Section provides definitions of quality measured in terms of latency and message loss. It also discusses efficiency in terms of link stress or minimizing the number of times a packet traverses an individual link in the network.

Results of studies of three real time simulation environments are discussed in Section 4. The propose of the studies was to develop a characterization of message flow in live simulations that could later be used to develop a reprehensive analytical model of the traffic flow and to influence the development of a proposed architecture for a new overlay protocol. In addition, the characterization of live message flows served to help validate the source traffic generator software module written to assist measuring the performance of the early XOM prototype.

An analytical model is presented in Section 5 that uses the information gained during the live simulation studies and experimentation in the laboratory. The proposed analytical model has a basis of metrics that are feasible to measure in the operational overlay network and therefore provides a relationship in a statistical sense to the overall expected performance of an overlay.

Section 6 provides a link between the analytical model and evaluation of an early prototype of the XOM overlay. The purpose is to validate the analytical model and serve to form a basis of key features in the proposed architecture.

Section 7 discusses the conclusions of the research and summarizes the unique contributions that resulted from the work. The section concludes with recommendations for future research that would support continued refinement of overlay multicasting.

SECTION 2 BACKGROUND AND SURVEY OF RELATED OVERLAY MULTICAST PROTOCOLS

I Introduction

There are many ongoing initiatives that are focused on the development of overlay multicast. These initiatives tend to all have roots in the context of satisfying a specific application environment in the same fashion this work is founded. This section begins with an overview of strategies for overlay multicast to establish a framework for the review of other overlay multicast initiatives that follow in part III of this section.

II Strategies for Overlay Multicast

Multicasting remains a critical element in the deployment of scalable networked virtual simulation environments. Multicast provides an efficient mechanism for a source of information to reach many recipients. Traditional multicast protocols, such as those defined by RFC 1075 [Wait88] and RFC 2362 [Estr98], provide mechanisms to support either one-to-many or many-to-many group communications typically associated with streaming media or distribution of large volumes data information. In real-time collaborative and virtual simulation environments, the requirement is for many senders to send to the same destination group(s) simultaneously. This is commonly referred to as many-to-many multicast [Moen03].

Even though IP multicasting was introduced nearly 20 years ago, it still is not widely available as an open Internet service even for one-to-many multicast [Deer90]. The most widely used multicast capability is the MBone [Erik94]. The MBone provides a circuit overlay inter-network that connects IP multicast capable islands by using unicast tunnel connections and is commonly used in university and research environments. Only recently have public carriers started to introduce multicast services, but then only as a private network offering where all interested parties obtain service from the same carrier [Wald03]. These new services are based on the more limited capability specified in RFC 3569 [Bhat03] providing one-to-many multicast.

Because open multicast services generally have not been available, there has been a shift to the idea of an end-host service to provide similar capabilities without involvement of the network provider. By organizing end hosts into an overlay to act as relay agents, multicast can be achieved through message forwarding among the members of the overlay using unicast across the underlying network or Internet. Two general approaches have been proposed to accomplish this. One is peer-to-peer networks that were originally designed for information sharing and messaging such as Napster and Gnutella [Chen00]. The second approach has focused on overlay multicasting to support group communications. Here, a transport-layer overlay, on top of the underlying Internet, between the members of a multicast group establishes group communications [Wang02].

The fundamental difference between these two approaches is that, in peer-to-peer networks, the topology tends to be random relative to the underlying physical topology which results from the loosely coupled relationship between the peers. The impact on the

service is that latency can be very high as information might pass across many peers some of which might be slow as well as have long physical paths between them in the underlying network. Also, large periods of message flooding can occur in peer-to-peer networks, this can cause congestion and inefficient use of network capacity. By contrast, an overlay multicast protocol can be more centrally controlled by managing the resources through a service node that can efficiently manage link stress defined as the number of times a message transmits across the same underlying network link.

The overlays are constructed under two different strategies: mesh or tree. The mesh strategy provides for more than one overlay path between each pair of nodes. In the tree case, a single path is established between any pair of nodes. It also is feasible to employ a mesh-first, followed by a tree construction algorithm, to implement overlay multicast where the idea is to take advantage of both strategies. In the following discussion, we will be concerned with failure/removal of overlay nodes. In most cases, the underlying network will include redundant paths.

There are distinct differences in the mesh and tree strategies that directly impact the control mechanisms of implemented overlay protocols. Tree overlays are sensitive to partitioning of the overlay because they are acyclic graphs. A graph that contains no simple cycles is defined to be acyclic where a simple path is a path that contains no repeated arcs and no repeated nodes, except the start node and the end node are the same [Bert98]. This means that if any non-leaf member of the overlay tree leaves the overlay, voluntarily, or by failure, the tree is broken and there will be no way for members of the multicast group to communicate until a new tree is constructed. The clear advantage of

trees is that, inherently, there are no routing loops formed during tree construction. This greatly simplifies the routing algorithm.

Mesh based overlays provide multiple or redundant connections between members of the group. This means that the overlay is less likely to be partitioned by node failure or departure. Alternate paths will already exist without the need to re-construct a path as is the case in a tree overlay. This certainly has advantages when considering needs for routing stability and offering quality of service (QoS) in the overlay. The down side to the mesh is that it is necessary to run a routing algorithm for construction of loop-free forwarding paths between group members [Wang02] such as a path vector algorithm. Mesh overlays also may result in some inefficiency as more than one copy of a message may use a link in the forward direction so that link stress increases. This is not the case in a tree nor is it necessary to run a routing algorithm once the tree is established in order to prevent loops, although the tree may be regenerated automatically as network conditions change.

Traditional tree approaches use core based or route point based approaches for forwarding messages. This approach works well for one-to-many multicast. The idea is that a sender that desires to send a message to the multicast group sends the message to the core of the tree or the route point node, which in turn then forwards the message along the tree to all receivers. There is some inefficiency that results because all sender messages must first be routed to the core or route point before distribution across the tree. Current IP layer multicast routing generally uses this approach. The network inefficiency can be overcome by using source based tree algorithms, in which each source builds the

optimal routing tree from the source to all receivers in the group, however, this approach results in more overhead as each node must now run a routing algorithm and maintain larger amounts of supporting information. However, storing and managing larger amounts of information is easier to accomplish on an overlay host than on an ordinary network router where processing and information resources per supported information flow tend to be more limited.

Another important aspect of overlays is whether or not they are constructed with knowledge of the Internet topology. An awareness of the underlying Internet topology improves the efficiency of the overlay. Data forwarding in overlay networks is done at the application level. Therefore, data may traverse the IP network several times before it reaches its destination or destinations. This may result in inefficient use of network capacity and increased delays compared to transmission at the IP layer. This disadvantage is reflective of all overlay protocols but is least pronounced if the overlay network is constructed with respect to the underlying Internet topology.

There are a couple of factors that influence the scalability of the overlay, where we define the scalability as the achievable size in terms of number of nodes or possibility in terms of overall performance, such as end-to-end latency. The number of nodes, for example, is influenced by the amount of information that a node might need to retain. If the information needs of a node grow faster than the number of nodes in the overlay, then this becomes the limiting factor. Limiting node information to only knowing neighbors, not the entire overlay, allows greater scalability. The level of effort required to build and maintain the overlay also can influence scalability. While processing power and network

capacity continue to grow, it is important to keep the overhead of the protocol in balance with the stated objective of efficient communications.

Typically multicast paths are the same as unicast paths and are the shortest paths in term of hops: end-host to end-host connections. The resulting shortest-path trees are good for best-effort traffic. However, when QoS is considered, such shortest-path trees may not have the resources to support the quality requirement. Therefore, it is desirable to include other resource availability considerations in the overall optimization of best path for offering QoS. Another example would be to use adaptive tree generation. This allows for traffic to move to the path with available capacity.

III Comparison of other Overlay Protocol Initiatives

Clearly, there are many alternatives and trade-offs for consideration in developing the optimal overlay multicast protocol, which are represented in a large number of initiatives in this area. Table A-2-1 below presents a summary of some of the more prominent initiatives that are in various states of experimentation and development. It seems that each of these efforts tends to focus on a specific optimization parameter that is reflective of a unique characteristic of a targeted application environment. While the intent of this research effort is not to draw a conclusion about this observation, it does however support the original proposal of this research. That is, that there are unique characteristics of the RT-DVS application environment that can be explored to enable open network overlay multicast services. The main characteristics of these applications are: real time, many-to-many, and receptive to network communication performance feedback. For example, RT-

DVS is unlike streaming video or streaming audio, which are also real-time applications where the sender is not necessarily network aware, but the transmission is one-to-many. Thus it is imperative to understand which combination of overlay strategies is optimal for RT-DVS such that the end systems cooperate to construct a good overlay structure to support many-to-many multicast.

To help answer that question, it is of value to review some of the most prominent efforts in overlay multicast protocol development. The row headings of the table indicate comparison criteria that reflect key performance elements for an overlay protocol. The criteria used for comparison are:

- Application: A general description of the targeted application environment, e.g. message information exchange, query, conferencing, streaming video.
- Overlay Topology: The term describes the nature of the organization of elements in the network. Examples would be, mesh, tree, ring, or multi-tier.
- Routing Algorithm: The routing algorithm refers to the specific algorithm used to develop the routing rules. Examples are distance vector, Floyd's shortest path, Steiner tree, etc.
- Group formation: A general description of how groups might be formed and managed in the overlay.
- Scalability measures: A description of the scalability of the protocol and measures used for determination.

- QoS considerations: A description of quality of services that might be offered or are part of the guarantee of the protocol. Included are considerations for priority, message loss, and path failure and recovery mechanisms.
- Consideration for Node characteristics/resources: Addresses whether the protocol considers the characteristics of a node in the development and dynamic management of the overlay. It is a recognition or consideration given to the ability of a node or host to act as an overlay relay agent.
- Node Join/leave/failures: Addresses the technique associated with nodes joining and leaving the network either by choice or fault.

In surveying these, our desired outcome is to identify techniques to support a protocol that is QoS sensitive even though the underlying Internet is not able to provide services at a consistent QoS [YanS02]. These comparison criteria are chosen as representative characteristics of a protocol that enable QoS sensitivity while being resource efficient and flexible. The criteria also represent areas or features that are typically traded off based on targeted application environment. In our RT-DVS application environment, the distributed real time simulation applications require a protocol that support many-to-many multicast while being sensitive to end-to-end latency [Moen03]. The environment must also be scalable to large number of users, which implies a protocol sensitive to efficiency or minimum control messages and flexible to manage many multicast groups.

Table A-2-1: Overlay Multicast Protocol Summary

Protocol	Characteristic	Comment
Narada [ChuU00]	Application	Not specific
	Topology	Mesh first. Begins with no knowledge of the underlying topology and over time continually refines the overlay as more information about the physical topology is obtained by probing
	Routing Algorithm	Distance vector Path algorithm on top of mesh
	Group Formation	Shares group information with neighbor nodes. The mesh creation and maintenance algorithms assume that all group members know about each other and, therefore, does not scale well to large groups.
	Scalability Measures	No consideration to node degree or link stress—it is what it is
	QoS Considerations	The mesh is dynamically optimized by performing end-to-end latency measurements and adding and removing links to reduce multicast latency
	Consideration for Node characteristics/resources	n/a
	Node Join/leave/failures	Randomly chooses nodes for consideration to join and over time continues improvement. Maintains time of last neighbor pulse/keep alive message in a queue and responds to time outs for discovering dead nodes/connections
Yoid [Fran00]	Application	Data gram or stream for Peer-to-peer
	Topology	Tree-mesh
	Routing Algorithm	Root based trees and includes formation of clusters
	Group Formation	DNS reference. Each group generates IP multicast address produced via a hash of the group ID and related information
	Scalability Measures	Hop by Hop transport. Unicast is used when more than one hop exists between nodes. Multicast is used in the LAN.
	QoS Considerations	n/a
	Consideration for Node characteristics/resources	n/a
	Node Join/leave/failures	Tree reconstruction

Protocol	Characteristic	Comment
oStream [CuiY04]	Application	On-demand media streaming (asynchronous)
	Topology	k-array tree
	Routing Algorithm	Minimal spanning tree-source based
	Group Formation	Based on request for like streaming media with adjustments to time of request
	Scalability Measures	Server capacity driven for buffering media
	QoS Considerations	Data buffering at relay nodes and at end host
	Consideration for Node characteristics/resources	Considers storage capacity for buffering
	Node Join/leave/failures	Tree joining is a local decision which implies only partial knowledge of the tree and recovery from node leaving is also local.
Mithos [Lieb02, Wald03]	Application	Peer-to-peer applications
	Topology	Geometric relationship based on distance where a unique ID is assigned that is used in the formulation of routing table. Geometries used could include rectangle, Delaunay triangle, quadrant-based, or closest to axis
	Routing Algorithm	Geometric. Measures distance to neighbors.
	Group Formation	n/a
	Scalability Measures	n/a
	QoS Considerations	n/a
	Consideration for Node characteristics/resources	n/a
	Node Join/leave/failures	Only interested in nearest neighbor. Uses geometric relationship
Tmesh [Wang02]	Application	Peer-to-peer
	Topology	Tree/mesh hybrid
	Routing Algorithm	Starts with an initial overlay and then uses Dijkstra shortest path first to discover shortcuts in the tree. For small number of members, not necessary to run a routing protocol.
	Group Formation	
	Scalability Measures	Adds shortcuts to improve tree delay performance
	QoS Considerations	End-to-end path delay measurements to improve tree performance
	Consideration for Node characteristics/resources	Discovers shortcuts in the tree such that efficiency and performance is improved.
	Node Join/leave/failures	Since it uses a mesh, needs only to recover from tree partitions

Protocol	Characteristic	Comment
Scribe [Cast02, CastUK]	Application	Messaging system for topic centric publish/subscribe messaging. Peer-to-peer
	Topology	Tree built on top of Pastry P2P network
	Routing Algorithm	Reverse-path forwarding Per group multicast spanning tree on top of Pastry nodes.
	Group Formation	Based on information topic subscription. Rendezvous point associated with a unique group ID. To create a group, a Scribe node asks Pastry to route a create message using the group Id as the key. The node responsible for that key becomes the root of the group's tree.
	Scalability Measures	Large numbers of members per group
	QoS Considerations	none
	Consideration for Node characteristics/resources	None. Relies on Pastry to optimize the routes from the root to each group member.
	Node Join/leave/failures	Keep alive messages to children. Message loss on failure of a single node. Local restoration of subscribers required in case of node loss.
Pastry [Rows01]	Application	Peer-to-peer
	Topology	Geometric (P2P) object location) based on nearest neighbor in terms of delay. Assigns a "proximity" metric that reflects the distance between a pair of nodes. Maintains a routing table with information about the distant nodes and a leaf set containing its direct neighbors.
	Routing Algorithm	Geometric based on nearest neighbors using unique node ID based on a secure hash
	Group Formation	Formation based on information topic subscription. Each group has a key called the group ID, which could be the hash of the group's textual name concatenated with its creator's name.
	Scalability Measures	Each entry in the routing table maps a node ID to its IP address. The routing table maintained in each Pastry node is created and maintained in a manner that enables Pastry to exploit network locality in the underlying network.
	QoS Considerations	None
	Consideration for Node characteristics/resources	None
	Node Join/leave/failures	Keep alive messages. Local restoration of subscribers required in case of node loss

Protocol	Characteristic	Comment
TAG [Kwon02]	Application	Applications with large numbers of members
	Topology	Tree
	Routing Algorithm	Each new member of a multicast session determines the path from the root of the session to itself, and uses path overlap information to partially traverse the overlay data delivery tree and determine its parent and children. The path computed under current Internet routing protocols serves as the basis for building the overlay network
	Group Formation	
	Scalability Measures	
	QoS Considerations	TAG constructs its overlay tree based on delay (as used by current Internet routing protocols), but uses bandwidth as a loose constraint, and to break ties among paths with similar delays
	Consideration for Node characteristics/resources	exploit the underlying network topology information for building efficient overlay networks where "underlying network topology," means the shortest path information IP routers maintain
	Node Join/leave/failures	Parent and its children periodically exchange reach ability messages in the absence of data. When a child failure is detected, the parent simply discards the child from its FT, but when a parent failure is detected, the child must rejoin the session
OMNI [Bane03]	Application	Single source media streaming applications
	Topology	Service provider deploys nodes in a network to act as overlay relay agents
	Routing Algorithm	Degree constrained average-latency algorithm results in directed spanning tree routed at source
	Group Formation	Initialization is a simple sort from the root based on latencies from the next node
	Scalability Measures	Degree bounded directed spanning tree
	QoS Considerations	Latency sensitive
	Consideration for Node characteristics/resources	Degree constrained and capacity constraints of the nodes.
	Node Join/leave/failures	Fixed nodes in the network that organize iteratively and adapt to changing network.

Protocol	Characteristic	Comment
Borg [Zhan03]	Application	Best effort delivery of peer-to-peer messaging
	Topology	Built on top of Pastry
	Routing Algorithm	Builds the upper part of a multicast tree using a hybrid of forward-path forwarding and reverse-path forwarding and leverages the reverse path multicast scheme for its low link stress by building the lower part of the multicast tree using reverse-path forwarding. The boundary nodes of the upper and lower levels are defined by the nodes' distance from the root in terms of the number of overlay hops.
	Group Formation	Creates a group by asking Pastry to route a create message using the group ID as the key for group formation.
	Scalability Measures	Large number of nodes. Uses Pastry network with average number of routing hops of at least 6.
	QoS Considerations	Best effort messaging
	Consideration for Node characteristics/resources	Dependent on Pastry
	Node Join/leave/failures	Node join/leave messages are sent towards the root using the multicast group ID.
QMRP [Chen00]	Application	Internet
	Topology	Tree and mesh. Uses single path and multi path to improve QoS
	Routing Algorithm	Behaves similar to PIM and is decentralized to local host
	Group Formation	New member inquires the session directory and sends a request message to the core. The message includes information about the QoS of the path.
	Scalability Measures	Maximum branching degree of 10
	QoS Considerations	Switches between single-path routing and multiple-path routing according to the current network conditions.
	Consideration for Node characteristics/resources	Detects termination of routing processes to improve responsiveness
	Node Join/leave/failures	Detects the failure as well as the success of routing without the use of timeout information.

Protocol	Characteristic	Comment
Hypercast [Lieb02, Hype02]	Application	One-to-many, Peer-to-peer
	Topology	Mesh based on hyper-cubes or logical Delaunay triangulations with source based trees for the data paths
	Routing Algorithm	Logical address encodes routing information from which the next hop information can be calculated without the use of a routing algorithm or table.
	Group Formation	
	Scalability Measures	Worst case is Delaunay triangle with 6 neighbors
	QoS Considerations	No QoS. Does not account for underlying network topology. The result is potentially large delay between node pairs
	Consideration for Node characteristics/resources	Only upper bound for number of neighbors-6
	Node Join/leave/failures	Node leaving requires more time for adjustment than a node joining
Overcast [Jann00]	Application	Single source streaming media content distribution,
	Topology	Tree
	Routing Algorithm	Overcast builds a single source-rooted multicast tree using end-to-end measurements to optimize bandwidth between the source and the various group members
	Group Formation	Single source multicast groups. Users join at a Overcast node
	Scalability Measures	Scalability of the root to handle large volume service requests
	QoS Considerations	Bandwidth
	Consideration for Node characteristics/resources	Bandwidth at the node
	Node Join/leave/failures	Maintains global status at the root of the distribution tree

Protocol	Characteristic	Comment
Tapestry [Zhao04]	Application	Tapestry is a peer to peer, wide-area decentralized overlay routing and location network infrastructure. Each node can act as a server to store objects, a router to forward messages, and/or client as source of requests
	Topology	Mesh
	Routing Algorithm	Hash-suffix mesh and allows messages to locate objects and route to them across and arbitrary network. Essentially, every node is the root node of its own tree which is unique spanning tree to all nodes.
	Group Formation	Unicast network overlay service
	Scalability Measures	Routing is inherently scalable
	QoS Considerations	Path is linearly proportional to the underlying distance.
	Consideration for Node characteristics/resources	n/a
	Node Join/leave/failures	Distributed mesh provides inherent alternate paths.
Bayeux [Zhua01]	Application	Steaming multimedia applications Peer-to-peer
	Topology	Dependent on tapestry
	Routing Algorithm	forward-path forwarding
	Group Formation	Uses session ID. A new request is sent to the root indicated session to join and root node manages the membership
	Scalability Measures	Root of the tree is potential bottleneck and single point of failure
	QoS Considerations	Relative delay penalty and physical link stress
	Consideration for Node characteristics/resources	n/a
	Node Join/leave/failures	Implements four control messages: JOIN, LEAVE, TREE, and PRUNE

A. Mesh Overlays

Overlay meshes provide the underlay that allow message forwarding mechanisms between members or nodes of the overlay. Essentially these meshes provide managed tunnels between nodes across the underlying IP network. Various strategies are

considered in performing establishment of these meshes including use of graphical shapes that have well known geometric routing principles as well as information about the underlying network or Internet.

Mithos [Wald03] uses a geometric approach where the network is embedded into a multi-dimensional space, with every node being assigned a unique coordinate in this space. The geometric approach greatly simplifies routing as routing is easily enabled with knowledge of the local grid coordinates. Hypercast [Lieb02, Hype02] also uses this strategy. This approach uses properties of regular geometric shapes like rectangles, hypercubes, or Delaunay triangles to greatly simplify routing tables. In fact, in the case of Hypercast, once the overlay is established, no routing protocol is necessary for the overlay.

In the rectangular approach, each node is assigned an enclosing axis-parallel multidimensional rectangle. Message forwarding is easily accomplished by sending to the rectangle abutting at the point where the vector to the destination intersects with the current node's rectangular boundary.

A Delaunay [Lieb02] triangulation uses the special characteristic that for each circumscribing circle of a triangle formed by three nodes, no other node of the graph is in the interior of the circle. Each node in a Delaunay triangulation has (x,y) coordinates which depict a point in the plane. This approach allows each application to derive the next hop forwarding information without the need of a routing protocol.

While Pastry [Rows01] is a peer-to-peer based protocol, the substrate self-creates a messaging routing overlay on the Internet that operates in a way that makes the overlay

look like a mesh. This is accomplished by each node having a unique 128-bit node ID. Using this unique ID, Pastry routes a message to the active node that is numerically closest. This approach provides a level of reliability since the idea is based on an active node. No further routing protocol is necessary for the local node to make this decision unlike a tree based approach were tree re-construction is likely required in the case of a node going inactive. Pastry also uses a metric for closest node such as latency so that optimum choice for forwarding is always made.

The HyperCast [Hype02] protocol builds logical overlays based on geometric properties of a logical graph. HyperCast currently implements both the hypercube [Hype02] and Delaunay [Lieb02] triangles. In each case, applications communicate with its neighbors in the geometric overlay, both in one-to-many multicast and many-to-one or anycast. The key advantage to using geometric logical relationships is that once the overlay is established, there is no further need for a routing protocol. The key disadvantage of this approach is that the underlying physical network is completely ignored, which makes it difficult to consider end-to-end latency in performance. Another disadvantage is that hypercube overlays must be formed sequentially with the result that for a large set of nodes, it is likely that it will take a long time to construct the overlay and also complicates departure or joining of a single node. In the case of Delaunay triangles, overlay construction can be accomplished faster since as they can be built in a distributed fashion.

The logical hypercube overlay network topology organizes the applications into a logical n-dimensional hypercube. Each node is identified by a label (e.g., "010"), which

indicates the position of the node in the logical hypercube. Message forwarding is easily accomplished by logical reference to nearest neighbor, hence no need for a routing protocol once the overlay is established.

Tapestry [Zhao04] is a unicast overlay network that provides the infrastructure for other multicast protocols like Bayeux [Zhua01]. The Tapestry routing mechanism is similar to longest prefix routing used in the CIDR IP infrastructure of the Internet [Rekh93], RFC 1518. The routing mechanism is a hash-prefix system which essentially results in every destination node being the root of its own tree that is the unique spanning tree across all nodes. The approach is inherently decentralized. Tapestry includes fault tolerant mechanisms that provide redundant paths to all destinations. This strategy effectively routes around failed nodes, in essence providing a mesh type infrastructure.

B. Tree Overlays

oSTREAM [CuiY04] is a tree based overlay that is specifically designed for one-to-many on-demand media distribution. The approach is to establish a minimum spanning tree and use media buffering at the host to aid in the distribution of asynchronous service requests for the same streaming media. While this strategy is not particularly useful for a many-to-many environment, there are some similarities in forwarding the same information to asynchronous requests in the web services model. This might apply, for example, in the case where more static background information such as terrain models or weather information might be asynchronously requested from group members in a

simulation. In these types of data distribution requests, there would be link and server efficiency advantages to using this approach.

Yoid [Fran00] employs a single shared tree for all members of the group. The links are unicast multi-router-hop paths. Trees are managed by a concept of child/parent relationship amongst members of the overlay. A member with no parent is the root member, and members with no children are leaf members and stub members are always leaf members. Each member divides the set of all other members into two groups called parent-side and child-side members. The groups are defined such that parent side members are all members reachable via the parent and all others are child-side members. Each member must mange this and understand how to protect from partition and make decisions on a new parent if the tree is partitioned.

Topology Aware Grouping (TAG) [Kwon02] exploits underlying network topology information to build efficient overlay tree networks among multicast group members. TAG uses information about path overlap among members to construct a tree that reduces the overlay relative delay penalty, and reduces the number of duplicate copies of a packet on the same link. Each member of a TAG multicast session determines the path from the root of the session to itself and determines its parent and children. TAG nodes need only the IP addresses and paths of their parent and children nodes. TAG is unusual in that it constructs its overlay tree based on delay and considers network capacity to break ties among paths with similar delays when constructing the overlay. This approach has merit for consideration as it provides the opportunity to guarantee end-to-end latency performance for the overlay. TAG does this by taking advantage of the underlying

network shortest path topology information maintained in the underlying network IP routers.

The Overlay Multicast Network Infrastructure (OMNI) [Bane03] is a two-tier approach to overlay multicast. The lower tier consists of a set of devices or service nodes that are distributed throughout an underlying network infrastructure like the Internet. The lower tier provides data distribution services to any host connected to an OMNI node over a directed spanning tree rooted at the source OMNI node. An end-host subscribes with a single OMNI node to receive multicast data service. The OMNI nodes organize themselves into an overlay which forms the multicast data delivery backbone. For the second layer, the data delivery path from the OMNI nodes to its clients is independent of the data delivery path used in the overlay backbone. This path can be built using network layer multicast, application-layer multicast, or a set of unicast paths.

Overcast [Jann00] is a single source multicast overlay designed for on-demand and live data delivery. The protocol is single source tree based with some added features to provide reliable delivery to multicast groups. Reliability is provided by using TCP e.g., HTTP over port 80.

C. Hybrid Mesh-Tree Overlays

As indicated earlier, there are two basic methods for construction of overlay trees for data delivery. First, one can construct a tree directly by members selecting their parents from among group members that they know. The second is to construct a well connected mesh of the group members and then use standard shortest path tree construction

algorithms to establish the minimum distribution spanning tree. Protocols such as Narada [ChuU00] apply a two step process where in the first step an overlay mesh is constructed and then a tree is constructed using a shortest path algorithm on the nodes of the mesh.

While the mesh construction can be accomplished in an arbitrary fashion relative to the underlying infrastructure, there is value in using knowledge of the underlying structure for building the mesh so as to improve overall performance of the overlay. The hypercube and Delaunay triangle techniques described above, for example, can easily be applied to build meshes without regard to underlying infrastructure. The technique might work well for peer-to-peer information exchange where end-to-end performance constraints are less stringent.

Narada [ChuU00], however, tries to build the mesh using knowledge of underlying network performance characteristics. Narada applies a reverse shortest path algorithm in the second step to establish shortest path minimum spanning trees with each tree rooted at the source node. Several advantages result from this approach:

- Group management can be accomplished at the mesh level and more easily allows for the use of a standard group management protocol like IGMP at the local distribution level.
- Meshes are more resilient than trees; repair and optimization are easier to accomplish as loop avoidance is not required during this process.
- There are many existing algorithms for construction of shortest path trees on top of the mesh.

Another protocol that uses this general strategy is Tmesh [Wang02], which uses an algorithm to determine shortcuts in the tree. The idea is to correlate measurable characteristics with a computed reduction in node-pair latencies attributed to each shortcut. This information is then used in a heuristic to select a shortcut with the objective of improving overall latency.

D. Peer-To-Peer Overlays

There is another definition of overlay networking that is normally associated with information or message exchange at the application layer without using the services of an intermediary host, such as in a client server application, called peer-to-peer. Peer-to-peer is typically used to define an end point or application layer exchange of information. They are not treated here as a separate category, as they apply routing principles similar to overlays in general, but the relationship requires some explanation as there are many developments in progress for implementation of self-organizing and decentralized peer-to-peer overlay networks [Zhan03]. These efforts support new distributed applications which require information discover and message exchange across a network of loosely coupled applications.

The point-to-point overlays typically implement distributed hash tables that allow for location of an object within a bounded number of routing hops. They tend to exploit proximity in the underlying network topology in locating objects and routing. Multicasting is built on top of these peer-to-peer overlays. Examples of this strategy include Borg [Zhan03] and Scribe [Cast02, CastUK] which are built on Pastry [Rows01]

and Bayeux [Zhua01] which is built on Tapestry [Zhao04]. Both Pastry and Tapestry provide unicast routing based on prefix-routing, and use a proximity neighbor selection mechanism to take advantage of the underlying physical network. Tapestry and Pastry also provide sensitivity to QoS by constraining the routing distance per overlay hop, resulting in efficient point-to-point routing between nodes in the overlay mesh.

Scribe uses reverse-path forwarding and Bayeux uses forward-path forwarding. The reverse-path construction of Scribe causes many short links in the multicast scheme which provides for lower link stress than Bayeux. Borg uses a hybrid multicast scheme to take advantage of asymmetry in routing in structured point-to-point networks. Borg builds the upper part of a multicast tree using a hybrid of forward-path forwarding and reverse-path forwarding and leverages the reverse-path multicast scheme for its low link stress by building the lower part of the multicast tree using reverse-path forwarding.

Peer-to-peer overlays networks generally are unpredictable and therefore impact quality of delivery of messages [Chen00]. Strategies such as message preservation priority in queues and expiration times are sometimes used to enable some level of QoS. An example is to discard messages that have reached expiration times, or at least give priority to those that have not.

IV Summary

The main objective of overlay multicast protocols is to provide a multipoint service that is independent of underlying network services. This allows unique applications such as distributed real time applications to be effectively and efficiently deployed across open

networks that do not provide a multicast service. This section has provided a summary of many initiatives designed to provide an overlay multicast service. These initiatives tend to be driven by unique characteristics of the target application environment. The result has been multiple strategies to achieve performance objectives for each of the unique characteristic environments. A similar challenge exits for the unique characteristics of the real time distributed simulation environment to find a solution to provide efficient multipoint communications services.

SECTION 3 BACKGROUND AND SURVEY OF RELATED ROUTING ALGORITHMS

I Introduction

Applying multicast routing algorithms in routing protocols is about optimizing certain objective functions to improve overall quality and efficiency of message delivery. Quality is typically measured in terms of latency and message loss rate. Efficiency is normally measured in terms of link stress or minimizing the number of times a packet traverses an individual link in the network.

The routing algorithms solve a shortest path problem, potentially with many constraints; to build a tree connecting the source(s) and destination group members so that minimal message flow occurs in the distribution of the message as well as maintaining optimum latency from source to each destination. The optimal case is that only a single copy of the message flows on any link in the overlay (link stress), meets application latency requirements (delay), and offers some level of reliability (minimum packet loss). Typical resources that are allocated or that must be optimized are link capacity, host processing capacity, number of links or diameter of the tree, and the degree or number of connections to other nodes in the overlay. In addition, the algorithm must lend itself to supporting dynamic overlays where many multicast group members join and leave in real-time. It is also important to consider the scalability of the algorithm which is made considerably more difficult in large networks since end systems in large networks

typically have limited topological information in which to construct good overlay paths. The underlying topology is also changing over time with new links joining and others leaving [Diot97].

The need for modeling and simulation multicast services can be defined as a multicast flow problem. This multicast flow problem involves sending flow from a source to a set of destinations or receivers. In this approach, a directed tree rooted at the source and reaching all receivers is used as the basic unit of flow for the multicast network flow. A multicast flow is then defined as a collection of flows along directed trees obeying the capacity constraints for each edge in a path. The general graph problem is normally defined as a weighted digraph $G = (V,E)$, node set V, edge set E, edge costs or capacities c_{ij}, source s and set of receivers R and includes a set of constraints as outlined above [Wang00].

If we use Wang's [Wang00] definition for M to be a multicast group or a subset of the nodes V that are involved in a group communications, we can then define a multicast flow as messages originating from a set of source nodes that are to be delivered to the set M destination nodes. These multicast flows can be one-to-one, one-to-many, or many-to-many. Our interest is in many-to-many as the most applicable to real time distributed simulations. The other cases can be described as sub classes of the more general class of many-to-many. In the many-to-many case, any node can be a sender or receiver, or both. Section 2 presented the general problem as a flow problem, repeated here for clarity:

> Given network $G = (N,A)$, node set N, arc set A, arc capacities c_{ij}, source s and set of receivers $R \subseteq N - \{s\}$, any multicast flow is identified with a

collection of directed trees $\{T_1 , T_2 ,\ldots,T_l\}$ rooted at *s* and reaching all nodes in *R* and associated flows $\{x_1 , x_2 ,\ldots,x_l\}$ with constraints:

$$\Sigma_{(1\le k\le l)}\, f(i,j,T_k)*x_k \le c_{ij} \quad \forall\ (i,j) \in A$$

$$f(i,j,T_k) = \begin{cases} 1 \text{ If arc } (i,j) \text{ appears in Tree } T_k \\ \text{and } 0 \text{ Otherwise} \end{cases}$$

The edge cost or capacity c_{ij} normally is associated with the expected delay a message might encounter on an outgoing link. This includes the queuing delay associated with the interface to the link, the capacity of the interface to the link, transmission time and the propagation delay along the link to the next node. The underlying routing algorithm would use the edge cost in an optimization to determine the multicast tree to be used to connect the source to all group member destinations.

As indicated above, many resources can be considered for allocation in optimizing the overlay. For real time modeling and simulation, the demand for improved quality of service over best effort implies consideration for other factors besides link delay in the construction of distribution trees. The desire is to have a "global" optimization such that the overall experience of the distributed simulation is consistent for the registered set of users. This is in contrast to a protocol like Overcast [Jann00] which optimizes network capacity rather than any consideration of other factors such as end-to-end latency or host resources.

Early laboratory tests of the XOM Relay (XOMR) prototype indicate the importance of knowing the resource capacity of the XOMR host. (XOMR refers to the actual protocol software.) In these laboratory experiments, the host performance in terms of

ability to service message interrupts was the limiting factor in performance. (The detailed results are presented in Section 6.)

If we assume that the access link capacity to the underlying open network or Internet is not the constraint, then the resource limitation of the host and end-to-end latency across the overlay are the constraints to be optimized. Studies of the three simulation experiments described in Section 4 support this assumption.

This appears to be a valid assumption for current simulation environments but is not necessarily valid for future applications such as tactical military networks, where access link capacity may be constrained. If we expand the application environment to other real time applications or extending simulation to operational environments, then the assumption of no access link constraint may not be valid. An example might be the use of simulation in a forward battle area of a military operation where network capacity is likely to be constrained. In this case, consideration to link capacity may become important as part of the overall optimization.

The traditional approach for management of link capacity constraints is through flow control. TCP provides this service in current reliable transport. The current XOMR prototype uses UDP but is not restricted to use of UDP and could be implemented using TCP. Since UDP provides no direct feedback mechanism for performance of a link, it then becomes more important for the XOMR to have some capability for link capacity constraint management to support potential applications in constrained link capacity environments. Simon [Simo04] proposed using a statistical approach similar to that used in the ATM protocol as way to predict forward congestion and then to implement

appropriate sending rate control at the sending host based on this prediction model. Other mechanisms such as reserving capacity or simply allowing only sending at a certain threshold rate could be used for rate control. This threshold could be a simple locally controlled capability based on known host resources.

Another possible approach implements class queuing at the sender host. In this approach, two classes of service, for example, might be offered to the application similar to how ATM provides different classes of service. In this case, lower priority traffic is discarded when there is projected congestion.

The remainder of this section provides a survey of possible routing algorithms and reviews them in the context of expected requirements of using multicast routing to support real time distributed simulations. This is an important differentiator as the opportunity in overlay multicasting is the ability to acknowledge unique features of the target environment.

II Algorithms Background

Multicast routing continues to become more important in distributed group communications and as a result has fostered a great many initiatives in developing new protocols and supporting routing algorithms. Diot [Diot97] provides a broad survey of protocols and supporting functions required for distributed group communications. The basic definitions associated with multicast routing provided in this reference are directly relevant and provide a very good basis for review of routing algorithms.

Reference [Diot97] summarizes the challenges of Internet multicast routing as the overall need to minimize network load, provide some basic support for reliable transmission of messages, and include the ability to design optimal routes under different cost constraints while minimizing the amount of statistical information stored at a router. These are well stated challenges. Similar challenges apply to overlay routing to support real time distributed simulation but with some differences in the objective functions.

Both Internet multicast routing and routing in general must avoid loops and have algorithms that provide for avoidance of traffic load on a link or subnetwork. This is also important in overlay multicasting, as the resources of the host and possibly the host access link are critical constraints in the overlay. The overlay assumes that the underlying Internet is providing optimal service using standard IP routing protocols.

Reliable transmission is aided by having stable routes. In IP layer multicasting this implies protection from link failures and awareness to minimizing route changes. In an overlay, failure translates to a node failure (host) either by the host failing or the host access link to the Internet goes out of service. Note that failure of node can include host overload or access link congestion. The overlay network then requires an assumption that the underlying network is stable and manages route changes internally that are invisible to the overlay.

One characteristic of real time simulations is able to help compensate for periods of congestion and possibly brief periods of lost messages either in the underlying network or in the overlay. This characteristic results from the design of these applications, as they have behavior algorithms that are able to predict the next state of a simulation object

without regard to receiving an update message, should the message be lost or not arrive in a timely manor. [Moen01]. This implies that the overlay performance has some flexibility in reliability that might not exist in other applications such as streaming video.

Another stated challenge is the need for minimizing state information stored in the routers of a multicast network [Diot97]. The concern in the overlay approach is not necessarily memory of the node host, but rather the amount of network traffic generated to distribute updates to this information in large networks. While available information is limited regarding how many participants there might be in a community of interest (COI) for a specific simulation environment, there is likely to be an expected limit on the number of players that could conceivably interact in a particular real time simulation. There are application environments, however, where distribution of real time information updates would be much different, for example an operational military environment.

The primary reason for routing algorithms is to design optimal routes, given a set of constraints. As discussed above, this normally translates to shortest path routes based on link or edge costs. There are other factors to consider in the overlay such as the resources of the host. We have already demonstrated this constraint in early laboratory tests [Moen04]. Overlay node access link capacity is another example and is complex in that the overlay host is likely to be sharing this link with other applications.

End-to-end delay is also an important constraint for real time simulations. This implies the introduction of path latency as a constraint in constructing the overlay. The real time simulation environment further refines this as a need for global optimization for latency. The reason for this as that these applications desire responsiveness measured in

human time. This means that all players in a COI have some expectation to be interacting in human time responses. For the players, interacting in different time frames does not have the same value and results in unresponsive behavior that might no longer be considered real time. The constraint can be defined as the diameter of the network where diameter means there is an end-to-end constraint that represents time or latency end-to-end across the overlay. This could be in terms of hop count if it correlates to time or sum of the edge costs of a path in terms of edge delay.

In light of these challenges, it is important to recognize that in the end, the routing algorithm must work with reasonable efficiency and also maintain a simplicity that provides ability to be managed and maintain stability for the supported applications. The paragraphs that follow provide an overview of a number of existing algorithms that are in various states of research and implementation. Each is viewed in the context of the merits to support real time simulations.

III Survey of Routing Algorithms

The simplest algorithm for multicast is broadcast or flooding the network with the messages to be distributed. This approach is obviously not at all efficient for large volume message distribution to large groups of recipients, however is very effective in local area networks and is the standard today for this application. It is simple and requires no calculations or algorithms to support message distribution in the local subnet. It is not effective in large groups of users distributed across many subnetworks.

Given the need for communicating with a multicast group and recognizing the need to optimize a set of resources or constraints, then the problem is to apply a routing algorithm that builds a multicast tree that optimizes an objective function. There are many proposed algorithms designed to solve this problem. Wang [Wang00] classifies multicast routing into 12 distinct definitions of the problem according to link and tree constraints. For reference, these definitions are quoted here:

"1) A link-constrained problem: A link constraint is imposed to construct feasible multicast trees (e.g., bandwidth-constrained routing).

2) A multiple-link-constrained problem: Two or more link constraints are imposed to construct feasible trees (e.g., bandwidth- and buffer-constrained routing).

3) A tree-constrained problem: A tree constraint is imposed to construct feasible multicast trees (e.g., delay constrained routing).

4) A multiple-tree-constrained problem: Two or more tree constraints are imposed to construct feasible multicast trees (e.g., delay- and inter receiver- delay-jitter-constrained routing).

5) A link- and tree-constrained problem: A link constraint and a tree constraint are imposed to construct feasible multicast trees (e.g., delay and bandwidth-constrained routing).

6) A link optimization problem: A link optimization function is used to locate an optimal multicast tree (e.g., maximization of the link bandwidth over on-tree links in a multicast tree).

7) A tree optimization problem: A tree optimization function is used to locate an optimal multicast tree (e.g., minimization of the total cost of a multicast tree). This is also known as the Steiner tree problem.

8) A link-constrained link optimization problem: A link constraint is imposed and a link optimization function is used to locate an optimal multicast tree that fulfills the constraint (e.g., the bandwidth-constrained buffer optimization problem).

9) A link-constrained tree optimization problem: A link constraint is imposed and a tree optimization function is used to locate an optimal multicast tree (e.g., the bandwidth-constrained Steiner tree problem).

10) A tree-constrained link optimization routing problem: A tree constraint and a link optimization function are used to locate an optimal multicast tree (e.g., the delay-constrained bandwidth optimization problem).

11) A tree-constrained tree optimization routing problem: A tree constraint and a tree optimization function are used to locate an optimal multicast tree (e.g., the delay-constrained Steiner tree problem).

12) A link and tree constrained tree optimization routing problem: Link and tree constraints and a tree optimization function are used to locate an optimal multicast tree (e.g., the bandwidth- and delay-constrained tree optimization problem).”

In the early description of the problem statement presented in [Moen03], it was defined as a classic Steiner problem because the objective is a global optimization of network resources. The global optimization here is defined as minimizing the total cost of the resulting multicast tree. By definition, if all the nodes are included in the group

membership, then this problem is the same as the minimum spanning tree problem. This is very a very attractive characteristic, one that should be considered in the architecture of the overlay protocol. While the Steiner problem remains a good description of the overlay multicast problem for distributed simulation, its solution is NP-complete [Wang00]. The same is true for problem definitions 4, 7, 9, 11, and 12 above.

According to Wang [Wang00], problems 1 and 2 are solvable because link constraints can be removed from the problems that do not meet selection criteria and can then be solved in polynomial time. Problems 3, 5, and 10 are solvable in polynomial time. This constrains the remaining possible algorithm approaches to those that consider single constraints or to use of an algorithm that is heuristic in nature so that at least approximate solutions can be obtained in polynomial time.

A literature search was conducted for proposed algorithms that would have potential for implementation in an overlay multicast network supportive of real time distributed simulation. The search focused on algorithms that were able to support source based routing and allow for consideration of constraints in addition to edge cost. The reason for this becomes apparent later in Section 6 which presents performance studies of the existing prototype multicast protocol that shows the importance of considering host resources.

Another important factor for consideration in identifying candidate algorithms is the expected size of the overlay in terms of number of nodes. The current prototype uses Dijkstra's [Corm97] shortest path algorithm for routing. It is a source based algorithm and performs adequately for the prototype overlay networks with small numbers of nodes

which have been part of early experimentation in this research, however may not be adequate for large overlay networks and the current version doesn't support multiple constraints. Dijkstra's algorithm also does not include consideration for other kinds of constraints, such as host resources without some modification to the algorithm [Boeh02].

A key assumption that is made for the overlay is that the underlying Internet does not represent a bottleneck in terms of capacity. This means that capacity measures in the routing protocol are not necessarily required across the Internet and algorithms that include network capacity considerations would not be required. However, as indicated earlier, this may not be the case for operational military networks.

The current approach for the overlay is based on the idea that this is a service provided by a host or a proxy on a Local Area Network that is available to all applications on the same subnet. The approach also recognizes that this is a service to a community of interest and therefore offering a host for this service implies a certain stability or availability of the host over the period of interest of the community of interest. What this means is that an assumption can be made that the overlay network is reasonability stable between networked overlay relay hosts. This raises the possibility that the problem can be simplified further.

Table A-3-1 provides a summary of algorithms that are being studied and evaluated by a number of researchers for use in overlay multicasting. The first column presents the name of the algorithm, though in some cases, this represents the name of a protocol as they are sometimes presented as synonymous. The source reference is also listed with the algorithm name.

Table A-3-1: Comparison of Routing Algorithms

Algorithm	Description	Objective Function	Complexity
DBDVMA – Distributed Bandwidth Delay Variation Multicast Algorithm [Huan03b,Rous97]	The algorithm constructs a widest available bandwidth tree under the constraints of end-to-end delay and inter-destination delay variation.	Optimize available bandwidth, end-to-end delay and inter destination delay variation.	Time: $O((\Delta-\delta)V\log V)$ Message: $O(V^3)$
STGM Shared Tree Group Multicast [FeiA01]	Builds shared tree by connecting group members one by one with bandwidth constraint.	Bandwidth constrained minimum cost path with ability to consider other QoS constraints though with increased complexity.	Time: $O(D(V+E)\log V)$ If a delay constraint or multiple QoS constraints are added, then the result is $O(DV^2E)$
BMSA Bounded shortest Multicast Algorithm [Pars98]	Two phased algorithm, first obtains a minimum delay tree using any well known shortest path algorithm like Dijkstra. Second, iteratively improve the cost of the delay-constrained tree. Assume knowledge of the complete topology at each node.	Minimum cost tree with delay constraints.	$O(kD^4\log(D))$
ALM Application-Level Multicast [Voge03]	The sending application assigns a priority to each pair of packet and receiver. The higher the priority, the more direct will be the path that the packet takes towards its destination.	Applies two steps for optimization, the optimization of resource usage leads to a MST, while the optimization of the cumulative end-to-end delay leads to an SPT.	

Algorithm	Description	Objective Function	Complexity
DDVCA Delay and Delay Variation Constraint Algorithm [Sheu01]	Combination of core based tree routing and the use of Dijkstra's algorithm for shortest path.	Minimize delay variation.	$O(DV^2)$
CST_{CD} and CST_C CST_{CD} uses both cost and delay and CSTc uses cost only to solve Constrained Steiner Tree [Komp93]	CST_{CD} chooses low-cost edges, but modulates the choice by picking edges that maximize residual delay where as the CSTc constructs the cheapest try possible while ensuring the delay bound is met.	Source based, delay constrained, Minimize cost and delay.	$O(V^3)$
Dijkstra [Wang96]	Shortest Path.	Shortest path.	$O(\|E\|\log\|V\|)$
Gallager's distributed algorithm for minimum weighted spanning tree [Gall83]	Constructs minimum-weight spanning tree in a connected undirected graph with distinct edge weights. Each node of the graph executes the algorithm, knowing initially only the weights of the adjacent edges. The nodes exchange messages with neighbors until the tree is constructed.	Minimum weight spanning tree with finite weight assigned to each edge.	Time: $O(V\log V)$ Messages: $O(5V\log V + 2E)$

Algorithm	Description	Objective Function	Complexity
K-SPH Kruskal-based shortest path heuristic [Baue96]	Distributed K-SPH starts with a forest of V multicast nodes and connects them pair-wise into successively larger sub-trees until a single multicast tree remains or no further connections are possible.	Node-degree constrained Steiner tree. Each segment determines closest segment with which to merge. Relies on shortest path information assumed available at each node.	Time: O(EV) Messages: O(VlogV)
SPH: Shortest Path Heuristic for degree-constrained Steiner tree problem (DCSP) [Baue95]	SPH initializes the multicast tree to an arbitrary multicast member. It then joins the next closest member to the tree.	Shortest Path tree construction beginning with an arbitrary starting point and an edge closest to the partial tree is added one at a time.	Performance curves provided in reference [Baue95]
Floyd [Floy62]	Finds all shortest paths in un-weighted digraph.h	Path weight is the number of edges in the path.	Time: $O(V^3)$
[Komp92]	optimal constrained approximation algorithm		
Distributed delay bound algorithm MST heuristic [JiaX98]	The construction of a routing tree starts with a tree containing only a source node. A destination in which is the closest to the tree is selected. The shortest path from the tree to this destination is added into the tree.	Network cost and bounded delay.	Time: $O(5V\log_2 V)$ Messages: $O(5V\log_2 V + 2\|E\|)$

(1) - Delay
(2) δ - Delay variation
(3) V – Number of nodes in graph
(4) D – Number of destination nodes in Tree
(5) k – number of shortest paths

IV Summary

Algorithms for constructing multicast trees seek to optimize the average of the minimum path delays from the source to each of the destinations and minimizing the cost of the tree where the tree cost is the sum of the cost of the edges of the tree. In the initial phase of this research, the problem definition used was the least cost tree approach which is defined as the Steiner tree problem. Unfortunately, the Steiner problem is NP-complete.

This Section presented a summary of heuristics and approaches for algorithms to approximate solutions to the problem. At one end of the spectrum complexity is the NP-complete Steiner problem and the other flooding a network with messages though obviously not very efficient in terms of link utilization. The ideal is to have an efficient algorithm that results in a multicast tree that spans the multicast group members that minimizes cost functions and minimizes the amount of information needed at each node.

SECTION 4 CHARACTERIZATION OF REAL TIME SIMULATION OFFERED MESSAGE LOAD

I Introduction

This section presents results from studies of three real time simulation environments. The propose of the studies was to gain an understanding of message flow generated by real time simulations. The primary objective was to develop a characterization or traffic model of the message flow that would assist in developing an Architecture for the XOM and result in the development of an appropriate analytical model for real time simulation message flow. In addition, the characterization of live message flows served to help validate the source traffic generator software module that was written to assist measuring the performance of the XOM prototype.

II Summary of Characterization Approach

The overall strategy for characterizing the message flow in a simulation environment was to instrument live simulation experiments. The OPNET modeling simulation software [OPNE05] tool set was used to aid in data capture and analysis. Multiple data samples were obtained in each of three simulation experiments. These experiments included:

- Wide area network consisting of 2 nodes distributed across the Internet using the XOMR prototype with each node running a Naval Post Graduate School

simulation of a maneuvering sea vessel. This experiment allowed for measurement of message flow for a single federate (object).

- Single simulation running on notebook computer as the host. This simulation was an Army ground operations maneuver to contact simulation where real-time was actual operations maneuver time or the same as wall clock time. This experiment allowed for instrumentation of an aggregate message flow in a controlled laboratory environment.
- Wide area network consisting of 3 nodes distributed across the Defense Research and Engineering Network (DREN). There were 211 Federates distributed across the network running an urban warfare simulation. This experiment allowed for the study of large volume message flow across an open network.

While all three where real-time simulation environments, there were significant differences in the characteristics among them which proved to be of great value in the overall understanding of the message flow in distributed real-time simulations. Two of the experiments offered unique opportunities as they were across open networks, therefore affording an opportunity to gather data in an environment where the XOM is expected to operate.

The one experiment that was conducted in the laboratory allowed for behavior analysis independent of any network layer traffic loading. This measurement experiment therefore offered the opportunity to view pure simulation generated traffic load, unlike the other experiments. This was of considerably benefit in defining an analytical approach.

Capturing performance information of networked simulations depends on complex interactions between the application, the services provided by the operating system of the Host, and the network layer services. Data capture must result in ability to provide detailed, quantitative understanding of these interactions across different network segments. The OPNET module called Application Characterization Environment (ACE) [OPNE05] was used to perform this data capture and analysis. The George Mason University C3I Center NETLAB was provided a license to use this product under the OPNET university research program specifically for this project.

ACE provides all the necessary capabilities to collect and analysis application message data across multiple segment network environments. ACE enables the capture, filter and the synchronization of multiple capture files. The capture files can be viewed individually or merged in order to provide an integrated view of message flow between elements of the environment being studied. The results are presented in graphical from and can present the end-to-end view of individual message flows. The capture files also can be exported to a text file where other approaches to analysis can be applied. All of these approaches were employed in capturing data and performing analysis of the three simulation experiments.

In order to perform a capture session, capture agents are installed on the targeted hosts of interest. The OPNET ACE central capture manager is installed on a central server that has network access to all the capture agents. From this central control module, all capture activities are controlled such as turning on all agents and then recovering data captured by each agent. This allowed for fully synchronized capture files.

Using ACE allowed visualization of application behavior at the individual transaction level which enabled graphical visualization and analytical representation of the behavior. The visualization is presented in a time synchronized, end-to-end view across a multi-tier network.

III Analysis of Naval Vessel Simulation across Open Network using Web Services and XOM Prototype

The objective in the first experiment was to understand offered network traffic load characteristics in an environment where multicast was employed. The sites involved were two academic XMSF laboratories: Old Dominion University (ODU) Virginia Modeling, Analysis and Simulation Center (VMASC), and GMU C3I Center. The RT-DVS workload consisted of the HLA Federation supported by the Pitch pRTI 1516, which is multicast-capable, and the Web Service Interest Management (WSIM) prototype as described in [Mors04]. The WSIM was used for a Web-service based viewer/controller for integrating and distributed simulation object update information from the source simulation applications to the simulation viewer clients at the remote locations. Simulated objects were naval vessels from the NPS SAVAGE library and visualized on a 3D viewer from NPSNET (http://www.npsnet.org). The simulation demonstrated no difficulty sustaining synchronized steaming-in-circles behavior with ship models running at ODU and GMU. The data rates involved were low: less than ten messages per second per site.

Figure A-4-1 presents the physical relationship for the sample network. The VMASC location is at Old Domain University Virginia Modeling Analysis and Simulation Center located in Suffolk, Virginia. The GMU C3I location is at George Mason University

Center of Excellence in Command, Control Communications and Intelligence Network and Simulation Laboratory located at Fairfax Virginia. Network communications occurred across Network Virginia, a regional part of the open Internet.

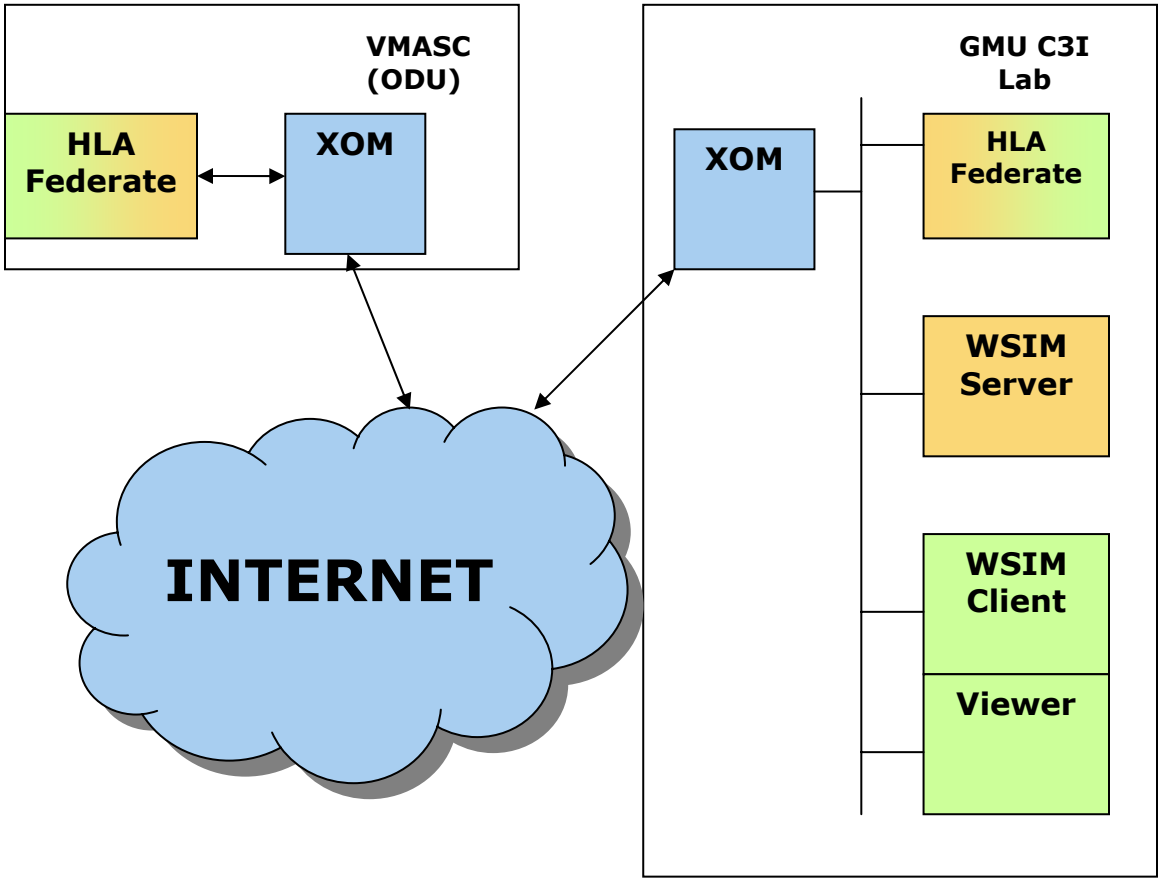


Figure A-4-1: Network Configuration for Web Service Interest Management

Figure A-4-2 indicates the logical tier relationships between the nodes in the network and the two multicast groups used in the message exchange. The tiers are defined as:

Simulation Federate, the WSIM Server (database and web services), Client, Viewer, and the two multicast groups.

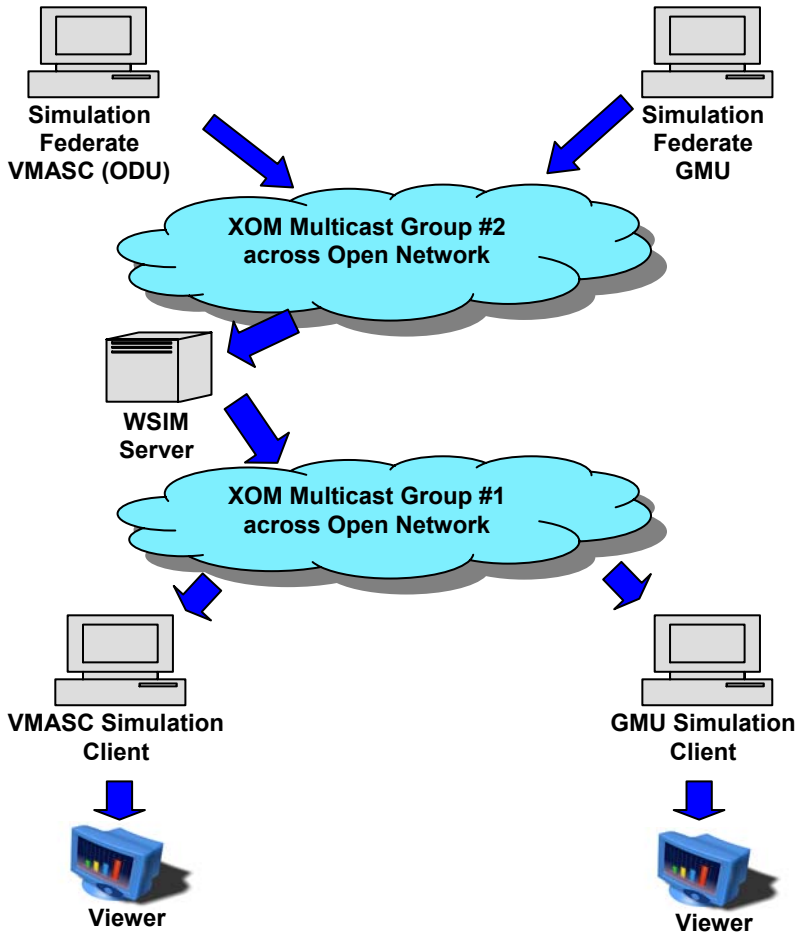


Figure A-4-2: Logical Tier Relationship for Multicast

The logical flow of messages is that the Federates publish to multicast group 2 using the XOM multicast services. The WSIM server listens to multicast group 2 and receives the published updates. The client, using TCP, establishes a tunnel connection to the WSIM Server announcing request for registered information and subscribes to multicast

group 1 using the XOM multicast service. The client then listens to multicast group 1 on the local area network subnet in the case of the GMU node and provides the received information to the local viewer. AT VMASC, the client listens to the same multicast group via the XOM services on the VMASC local area network subnet.

For network performance data capture, we used the OPNET ACE application hosted on a separate server at the C3I NETLAB. ACE capture agents were placed on the targeted hosts for data capture and were controlled by the OPNET ACE host. Figure A-4-3 graphs the message throughput from a single Federate to the WSIM server.

In our sample experiment, the naval vessel simulator Federate represents a single simulation element or object. This allows for excellent understanding of the value of the multicast service as it relates to a single object in a visual simulation. It is easily observed in Figure A-4-3 that this particular simulation object generates uniformly distributed message flow at the rate of 4 messages per second with a periodic message that starts a message transfer for an object update. This message rate is consistent with our early model development for simulation objects [Moen01]. The simulation object is in a continuous movement state, thus generating uniform update messages at a uniform rate. In reference to our earlier model, the object essentially never changes state from the "move" state.

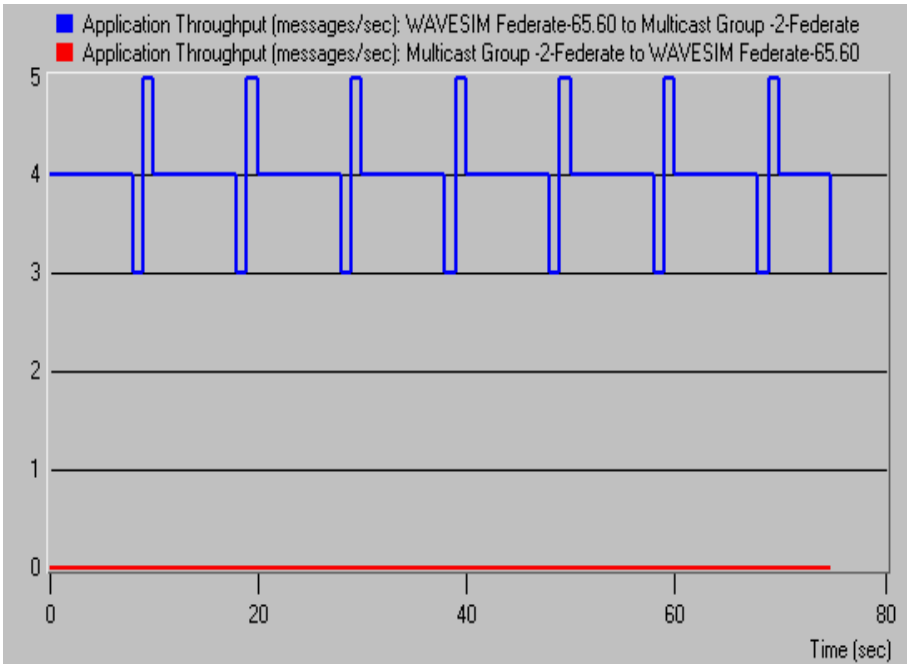


Figure A-4-3: Message Flow from Federate to Multicast Group 2

Figure A-4-4 provides another view of the message flow, based on a measure of the inter-arrival times of the messages. The x-axis represents an individual message arrival in sequence of arrival over time. The y-axis represents the inter-arrival time of the message in seconds. The figure shows an inter-arrival time centered on .25 seconds which by inverting gives us the 4 message per second rate. The standard deviation for the inter-arrival times is .04228 which supports the observation of a deterministic distribution.

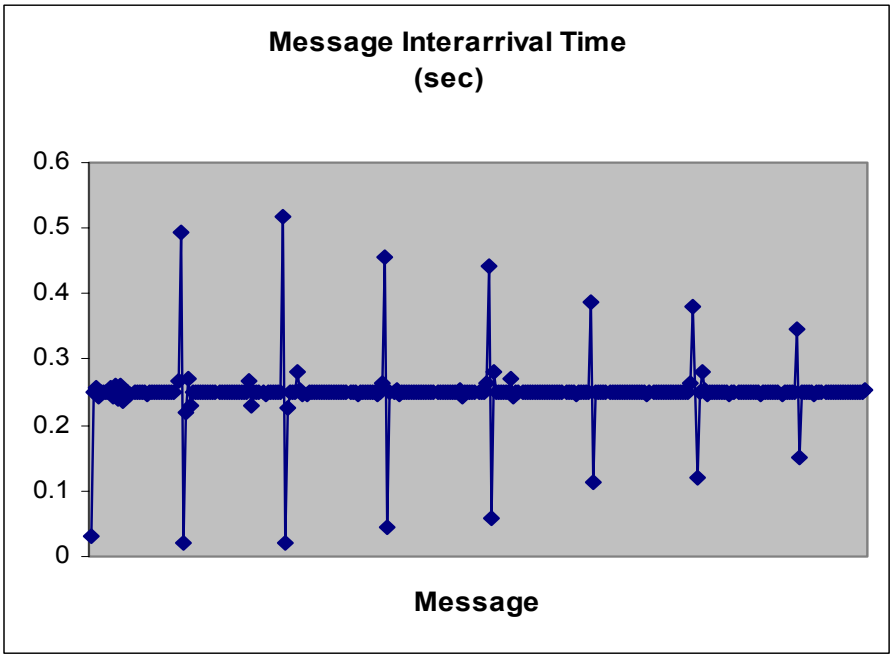


Figure A-4-4: WSIM Message Inter-Arrival Time

Recall that this experiment used the WSIM which provides a Web service that integrates and filters information exchange between multiple simulations. In our early experiments using the NPS vessel simulation with the WSIM services [Mors04], we relied on these message transfers using TCP connections. This entailed a TCP connection for each message and each tier in the information exchange. With the TCP connection, each message transfer generated 2 packets at the network layer for each message for every element of the transfer tier. Whereas using the XOM multicast service, a message transfer entails a single packet at the network layer therefore significantly reducing the overall network load.

In addition, the multicast provides reduced network link stress in that there is a single message transfer between XOMs for a multicast group with the message available to any

user on that local subnet. With multiple sites, the advantages of multicast would have been even more pronounced.

We also simultaneously captured message flow data from the WSIM server to Multicast group 1, the integrated message flow group the client viewers received. This message flow integrated the simulation update messages from each of the single vessel simulations into an integrated view of all the vessels for the client viewer. The viewer then displayed all the vessels as an integrated simulation visualization even though the source data where separate single vessel simulations. The results of this capture are indicated in Figure A-4-5. Here we observed a slightly different message flow pattern, one that reflects an ON/OFF pattern where OFF implies a lower transmission rate not necessarily zero. Notice that the periods, or length of time that messages are in the OFF or ON state appear to be vary slightly over time. By inspecting the raw data, the ON period represents the transmission of 2 or 3 messages at a deterministic high rate averaging 8 messages per second with a periodic message transfer at a higher rate followed by one at a low rate.

The pattern is similar to the Federate uniform rate but periodically has these minor rate changes. One possibility for this pattern is the random arrival of the second Federate message relative to the first as a result of wide are network delay and the effect of processing delay associated with the arrival before the WSIM aggregation message is forwarded to the multicast group.

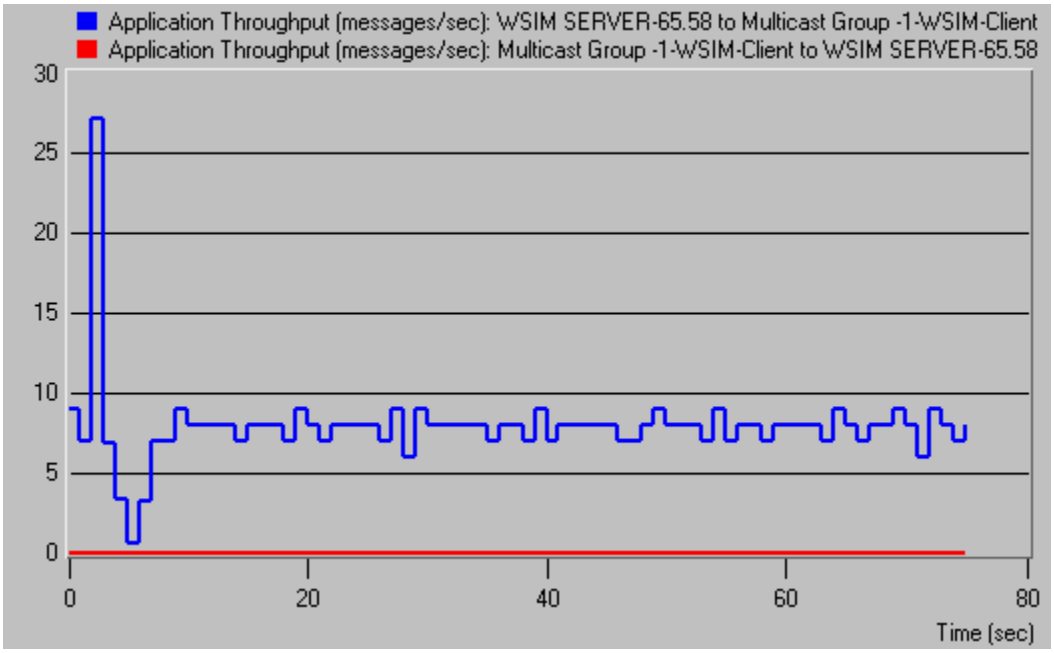


Figure A-4-5: Message Flow from WSIM Server to Multicast Group 1

To investigate this pattern, we used the OPNET ACE feature to look at the data by message from a tier flow perspective. Figure A-4-6 presents this view. This Figure displays the overall flow of application-layer data between tiers where each arrow represents a single message. The specific data capture interest is the flow from the WSIM server to multicast group 1. Processing of arrivals from each Federate at the WSIM server spawns a message to multicast group 1.

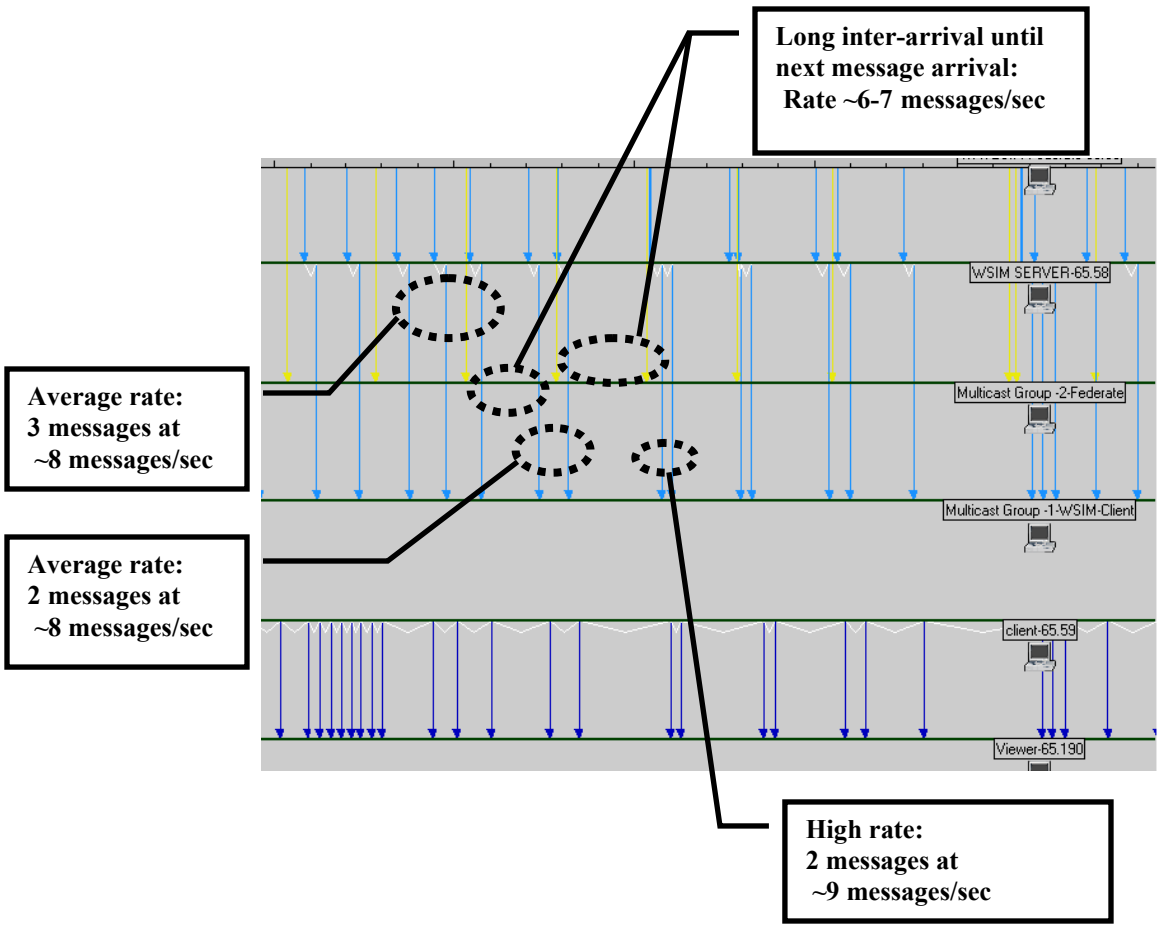


Figure A-4-6: Tier View Message Flow

The figure conventions are:

- Time is marked in seconds along the top.
- Each solid horizontal line represents a tier (host) (labeled on the right).
- Each arrow represents a single application message. The locations of an arrow's head and tail represent the message departure and arrival time.
- Application message groups represent closely spaced application messages.

- Dependencies indicate network or application delay. A V-shaped line beneath a tier indicates delay between two messages at that tier.

Another observation is that the message flow rate is never zero for any extended period of time which supports the idea of ON/OFF pattern implying high and low rates of message arrival. This observation is an important consideration as we develop a class of service (priority) capability for the XOM. A continuous message flow rate in a priority queuing system implies that the possibility exists that a lower priority arrival will never receive service unless a form of weighted fair queuing is implemented.

These observations also impact global optimization of the overlay resources such as the network access capacity for each of the XOMRs, link capacities, and number of intermediate XOMRs in an overlay. This experiment also demonstrated the use of multicasting as a mechanism to improve efficiency in the distribution of integrated simulation information across an open network.

IV Ops Simulation

The second simulation experiment was a real-time combat unit operations maneuver simulation consisting of a background static terrain map overlay and approximately 30 mobile/active objects. The general description of the simulation is a friendly force combat unit maneuvering to engage an enemy force deployed in defensive positions. Each object represented a mobile weapons system in the case of the friendly forces and a defensive position with a weapons system in the case of the enemy forces. Real time in this simulation is defined as actual operations clock time. This requires many hours of

real clock time to completely play out the operations scenario, therefore, the total scenario was not played to completion but only through initial deployment of friendly forces upon first contact with the enemy forces.

The data capture was accomplished by running the simulation on a notebook computer with a crossover cable connection to the ACE central host. There was essentially no network in the loop. This is observed by the simple message flow diagram in Figure A-4-7. The diagram indicates message flow in the direction of the arrow with the total number of messages indicated by the arrow call-out (18359 messages outbound from the OPS-SIM Host and 0 messages in return). Twelve Data sample collections were made, each approximately 200 seconds of real clock time. Results from only a single representative sample of a data capture are presented here. There were no observed significant differences in the separate data capture files.

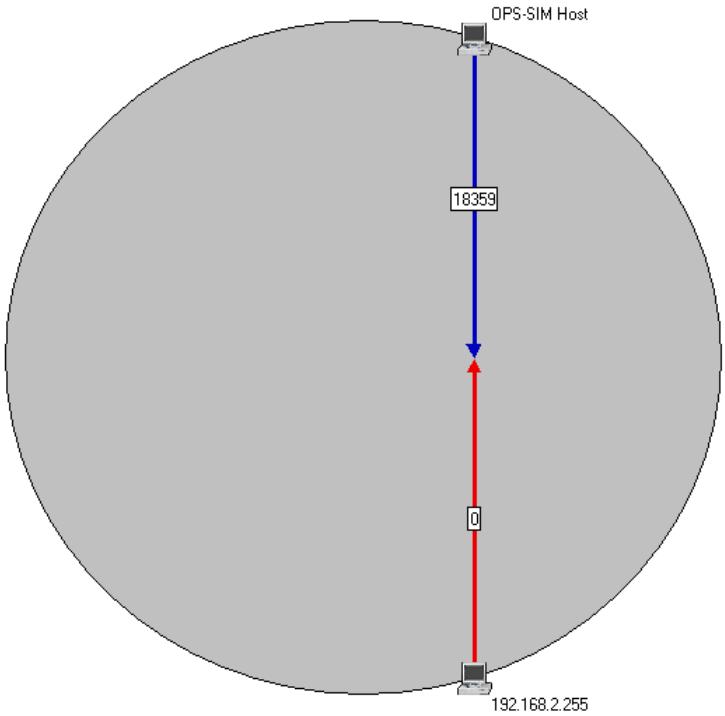


Figure A-4-7: Operations Simulation Experiment

The test configuration was very simple with the simulation host connected to the capture agent host using a cross over cable between the two hosts providing standard Ethernet network capacity of 100 Mbps. There were no other network connections or other sources of message traffic. There were no observed delays for networking which is what was expected for this network connection approach and relatively low traffic volume.

The transaction has 2 tiers: operational simulation (OPS-SIM) host at 192.168.2.51 and the Ethernet multicast address at 192.168.2.255. Figure A-4-8 displays the overall

flow of application-layer messages between tiers focusing on the time interval 126.7 seconds to 127.3 seconds.

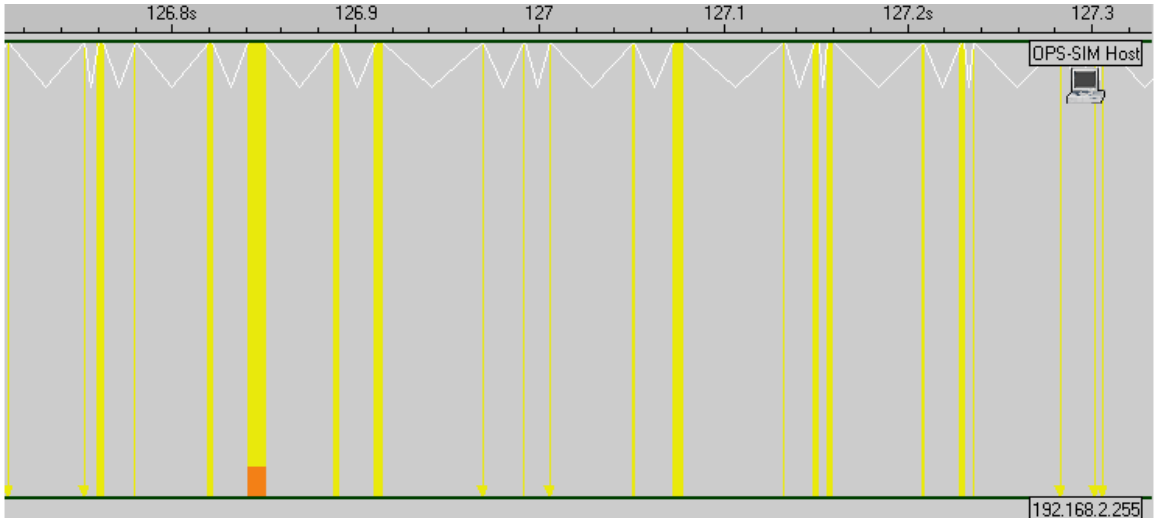


Figure A-4-8: Tier View of Message Flow

The most important observation to be made by from this chart is that there are periods of time during which there is a much higher rate of message transfer which is observed by the message groups represented by the vertical bars rather than a single arrow. The largest occurs at approximately 126.84 seconds into the data capture window. Also, there are a number of smaller, but similar in size groupings appearing at 126.89, 126.91, 127.07, 127.15, 127.16, and 127.23 seconds. This pattern could be described as an ON/OFF pattern where ON means a higher rate of message transfer and OFF means a lower rate, not necessarily no message transfer.

The second important observation from reviewing the detail data is that proportionally, the message sizes tend to be greater than 100 bytes. This is expected for

this application as the standard message applications size is approximately 150 bytes. Only during the largest group do we observe a small amount of messages with size below 101 bytes indicated by the different color shade at the bottom of the bar at 126.84 seconds. The ON/OFF message pattern is more easily observed in Figure A-4-9 below which shows the average number of application messages transmitted.

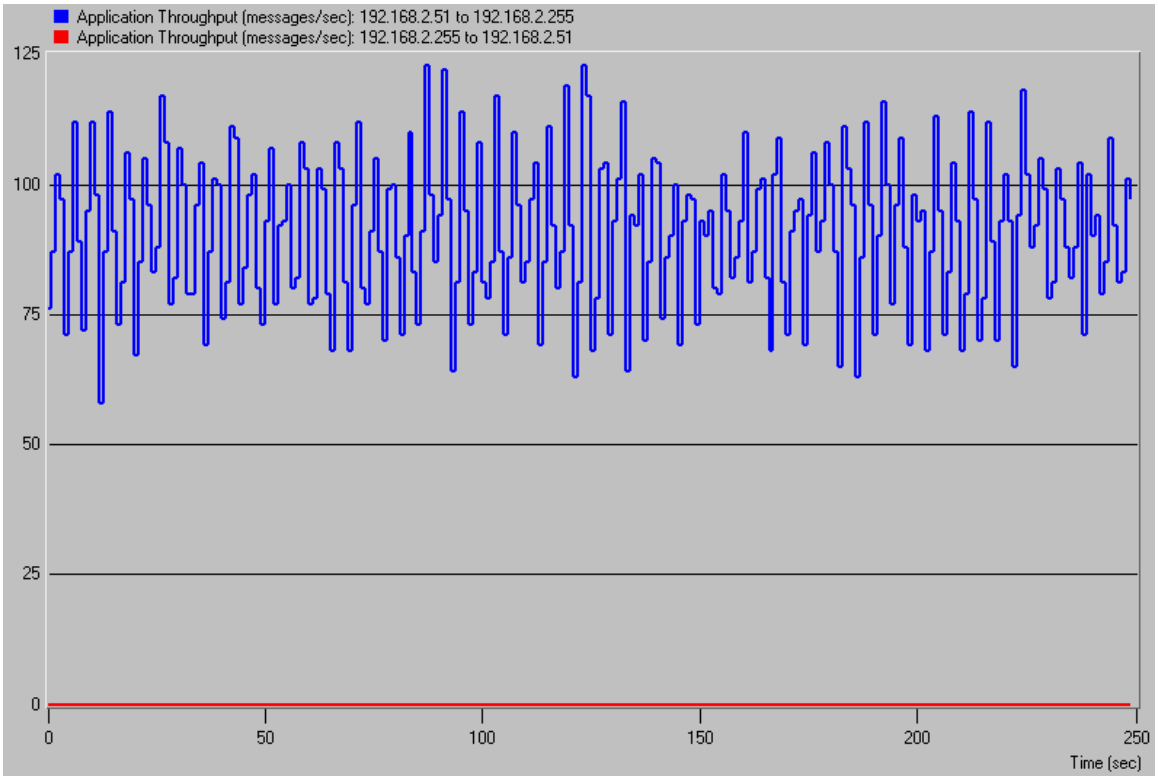


Figure A-4-9: Message Throughput for the Ops Simulation

The message inter-arrival time is presented in Figure A-4-10 below. Analysis of the detailed data in the capture file reveals an average message inter-arrival time of .011008 sec which translates to an average of 90.9 messages per second.

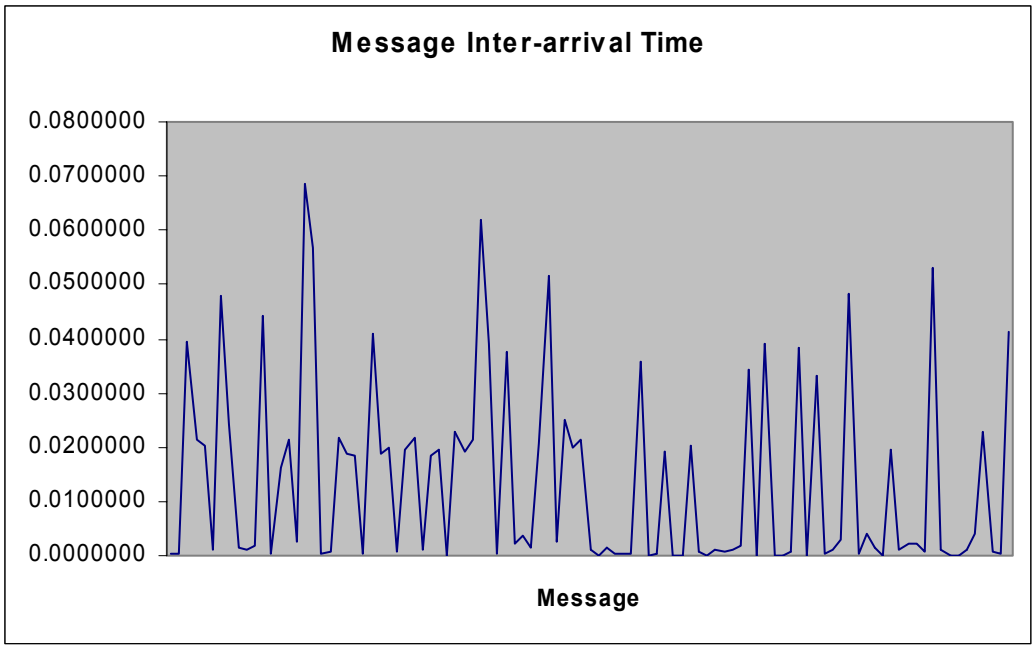


Figure A-4-10: Ops Message Inter-Arrival Time

If we discount the bursts (peaks) both high and low, then it is observed from the graph that the ON periods have message flow rates of approximately 100 messages per second and the OFF periods have message flow rates of approximately 80 messages per second. However, inspection of the individual message arrival times indicates that the rate of message arrival varies over these ON/OFF periods and appears to be random.

If we apply our earlier estimate of approximately 30 active objects in the simulation, then we have an average of 3 messages per object per second with ON/OFF rates of 3.33 and 2.67 respectively. These rates are similar to the original test traffic generator model [Moen01] which has an average of 3.1 messages per object per second.

Finally, analysis of the data capture file reveals an average message size of 145 bytes. The minimum size observed is 72 bytes and the maximum size is 192 bytes. The standard

deviation for the message size was calculated to be 35.17. The network protocol overhead was observed to be 42 bytes for all messages, which is what was expected. Therefore, the average network packet size is 145 bytes + 42 bytes, or 185 bytes. Average inter-arrival time is .0110008 or 90.9 messages per second.

V JFCOM Message Transaction Analysis of JFCOM Simulation Experiment

As part of a larger research effort funded by the Defense Modeling and Simulation Office (DMSO), George Mason University C3I Center participated in a JFCOM J9 simulation experiment during the period October 13, 2004 to October 15, 2004. The participation allowed for the characterization of message flow between elements of a large distributed simulation experiment.

The wide area network had three nodes connected by high speed access (OC3) to the Defense Research and Engineering Network (DREN). The nodes were located in Northern Virginia (Ft Belvoir), Southern Virginia (Suffolk) and Southern California (San Diego) as indicated in Figure A-4-11. The DREN is an open network environment similar to the open Internet using standard Internet protocols for routing and message transfer.

Distributed across these three nodes were 211 federates connected via Run Time Infrastructure (RTI). The simulation environment contained approximately 110, 000 entities or objects. All communications between federates used reliable communications by TCP. No multicast was employed. However, a “pseudo” form of multicast was employed by using intermediate hosts called IMP’s for aggregation of object update

messages. (IMP is an acronym used to describe the background clutter used to simulate an urban area.)

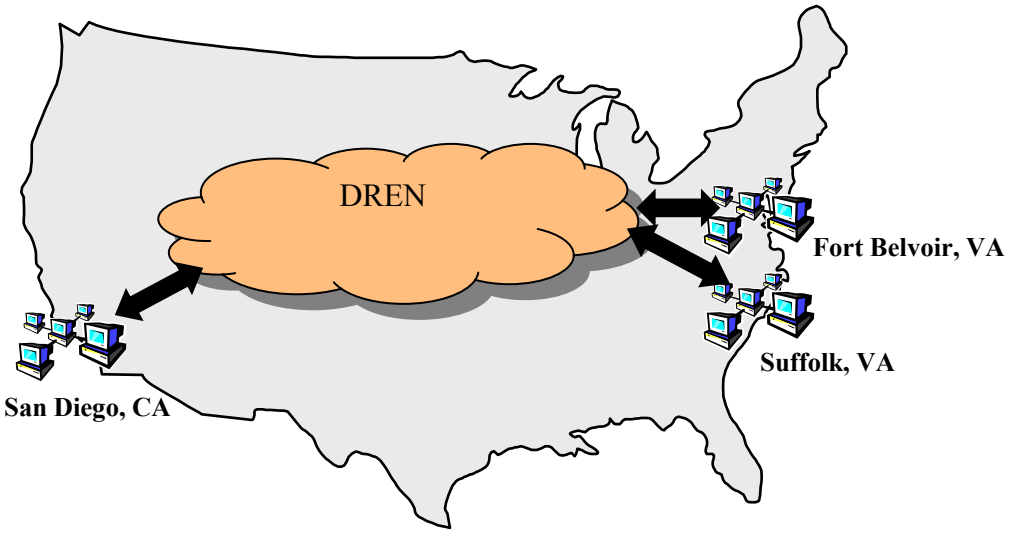


Figure A-4-11: Distributed Network

In previous experiments operated in the same environment, attempts where made to use standard Internet multicast protocols rather that then intermediate relay host strategy. However these protocols proved too complex and difficult to manage. They also were reported to not meet the reliability and other performance needs of the simulation environment. Specifically, the management of group membership did not easily support the nature of the applications need for joining and leaving group subscriptions.

The alternative was implementation of "pseudo" multicast. The update message concept employed was a publish-subscribe model where the aggregate visual space was divided into a fixed grid overlay. A group number was assigned to each grid in the overlay and the objects with in the grid where assigned to the associated group number.

A federate subscribes to a group(s) as necessary based on desired view of the overall visual space. The information is then distributed from the head Federate (head IMP) across a hierarchical structure of IMP hosts. This concept is presented in Figure A-4-12. Notice that at each layer, there can be several instances of a lower layer host, providing a complex distribution scheme. The number to the right of the brakes in the figure indicates how many times each layer host is replicated.

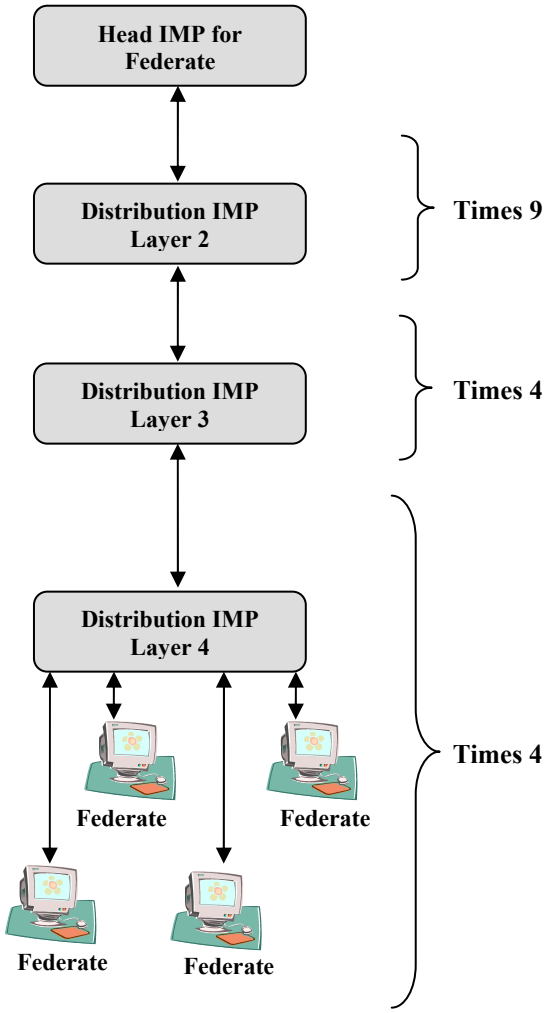


Figure A-4-12: Typical Hierarchal Distribution

The IMPs in the structure are responsible for store and forward of element message updates based on group subscriptions of associated Federates. The service provided by this approach makes no assumptions or guarantees about end-to-end latency for the delivery of messages. In the case of this experiment, the users accepted the inherent delay and learned to adjust their interactive response to the delay during the course of running a real-time operational simulation. In other words, the users accepted the response times of the longer delays.

Multiple traces of date where collected over a two day period of the experiment. The following data summary is based on information from one trace of length 8.6 seconds collected at a Layer 4 IMP labeled Blue IMP-163. The Figure A-4-13 below presents a tier view of the studied trace. Notice the source message flow is from a Layer 3 IMP labeled Blue IMP-162. The numbers in the rectangular boxes indicated the total number of messages sent during the collection period with the arrow heads indicating flow direction.

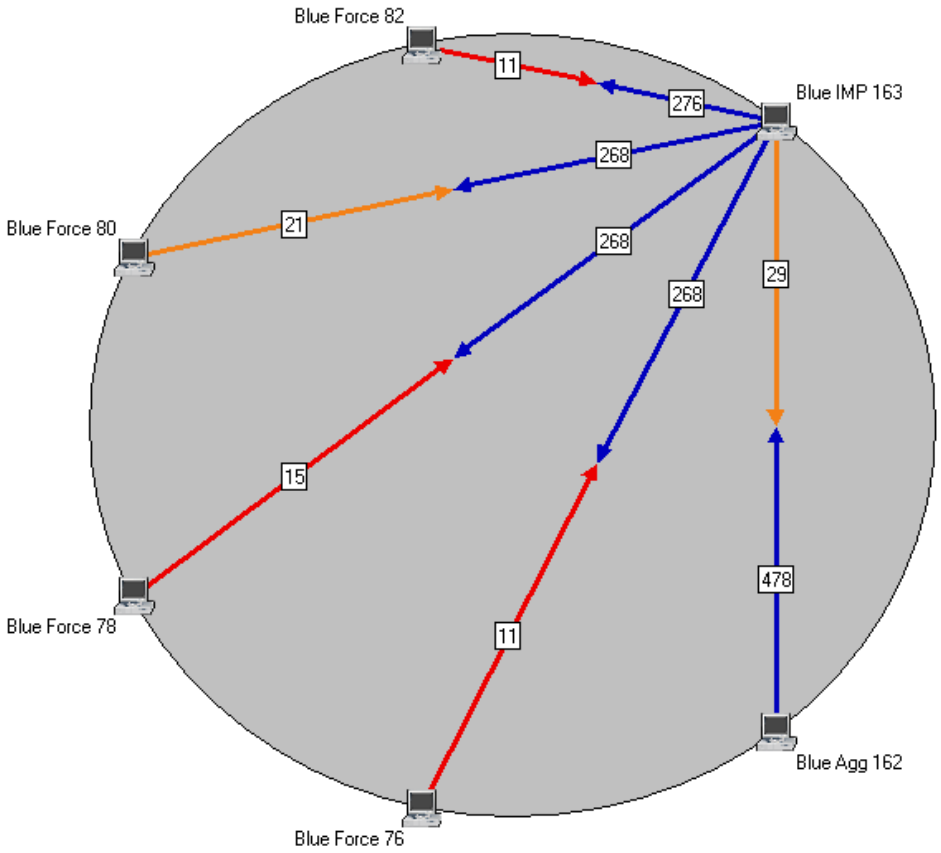


Figure A-4-13: Tier Relationship of Message Flow

It is also interesting to look at this flow from the amount of data flow. Figure A-4-14 below presents the same traffic flow but in terms of bytes. It is more telling in terms of the re-use or relay of the same information. The sum of the total number of messages from the Blue IMP 163 to the four Federates indicated in Figure A-4-13 is 1080. Dividing by the total message flow from the Blue Agg 162 IMP implies a re-relay rate of 2.26 times per message arriving at Blue IMP 163. If we consider the same message flow in terms of bytes of data (Figure A-4-14), then we have a relay rate of 4 times the amount of

data arriving. The implication is that the Layer 4 IMP is processing the arriving data and re-aggregating or re-arranging the data flow into messages to the subscribing Federates based on specific requests. Since the Federates are all local to the Layer 4 IMP, there would be some efficiencies to be gained by implementing a multicast service with efficient group management.

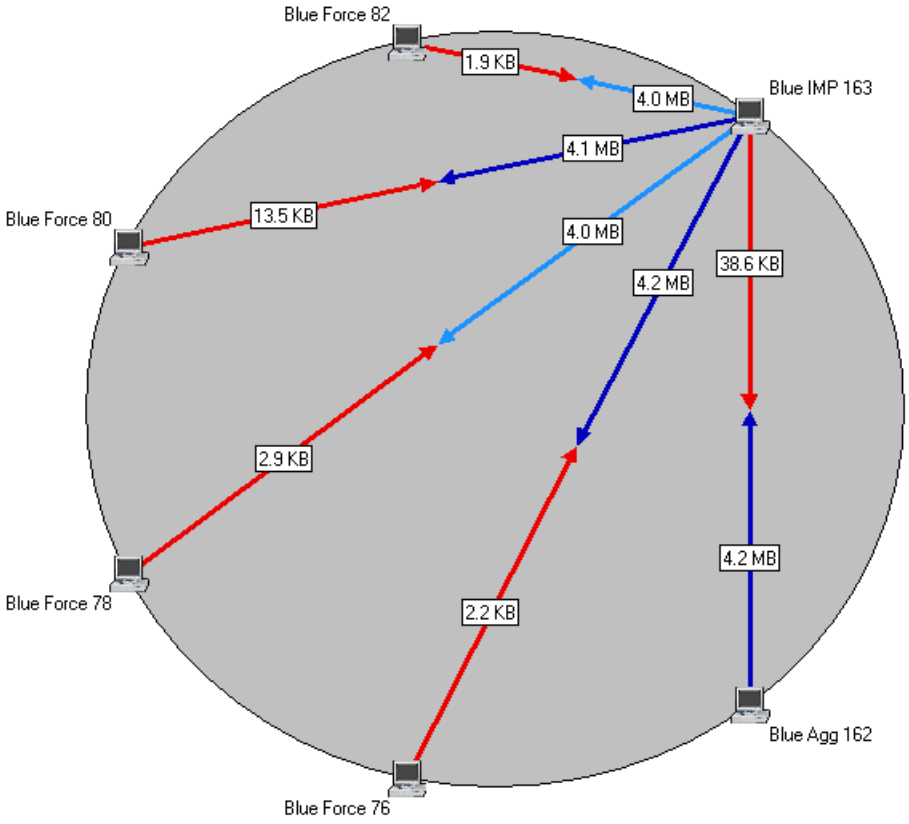


Figure A-4-14: Tier Relationship of Message Flow (Bytes)

A complete summary of the data flow is presented in Table A-4-1 below. Notice that the average size of the application message is quite large, on the order of 9 to 15 Kbytes. This translates to a very large number of network packets.

Another interesting observation in this table is that only one message was retransmitted as a result of an out-of-sequence condition. This represents only a .06 percent retransmission rate for this sample. This result is similar in all the other data samples. Given this very low retransmission rate, it implies that using UDP rather than TCP would result in a significant improvement in efficiencies for this distributed simulation for this network configuration. Using UDP instead of TCP would reduce message load on the network because of the reduced protocol overhead of UDP.

Table A-4-1: Summary Statistics – JFCOM Simulation Experiment

	Across All Tier Pairs	**Blue Force 76 □ Blue IMP 163**	**Blue Force 82 □ Blue IMP 163**	**Blue Force 80 □ Blue IMP 163**	**Blue Agg 162 □ Blue IMP 163**	**Blue IMP 163 □ Blue Force 78**
Response Time (sec)	8.590860	8.588108	8.583471	8.583341	8.584395	8.583232
Application Turns	136	14	14	32	54	22
Application Messages	1,645	279	287	289	507	283
Application Data (bytes)	21,435,132	4,390,236	4,149,524	4,288,261	4,437,625	4,169,486
Average Application Message (bytes)	13,030.48	15,735.61	14,458.27	14,838.27	8,752.71	14,733.17
Network Packets	19,432	3,716	3,540	3,691	4,947	3,538

	Across All Tier Pairs	Blue Force 76 □ Blue IMP 163	Blue Force 82 □ Blue IMP 163	Blue Force 80 □ Blue IMP 163	Blue Agg 162 □ Blue IMP 163	Blue IMP 163 □ Blue Force 78
Network Data (bytes)	22,719,684	4,635,524	4,383,196	4,533,779	4,764,159	4,403,026
Average Network Packet (bytes)	1,169.19	1,247.45	1,238.19	1,228.33	963.04	1,244.50
Max Application Bytes Per Turn (A □ B)	Not Applicable	1,298	856	2,896	664,017	1,566,367
Max Application Bytes Per Turn (A □ B)	Not Applicable	1,962,405	1,911,113	772,447	6,660	1,334
Retransmissions	1	0	0	1	0	0
Out of Sequence Packets	1	0	0	1	0	0

Inspection of the source message traffic indicated that the dominant traffic source to be the scenario geographic background generator connected to the Head IMP. This message traffic was referred to as “clutter” as it represented the real-time simulation of an urban environment including normal vehicular traffic and movement of pedestrians. The federate was observed to generate a very low volume of message traffic and in most cases was ACK messages associated with TCP message flow.

If the message flow data is plotted as the average message flow rate over time, the result is that presented in Figure A-4-15. This figure is an integrated view of the flow where the flow is from the Layer 3 IMP to the Layer 4 IMP and finally to the Federate. The upper line on the graph is the higher layer flow and is nearly twice the rate as that

between the lowest layer and each of the Federates as indicated in the tightly coupled set of lower lines on the graph.

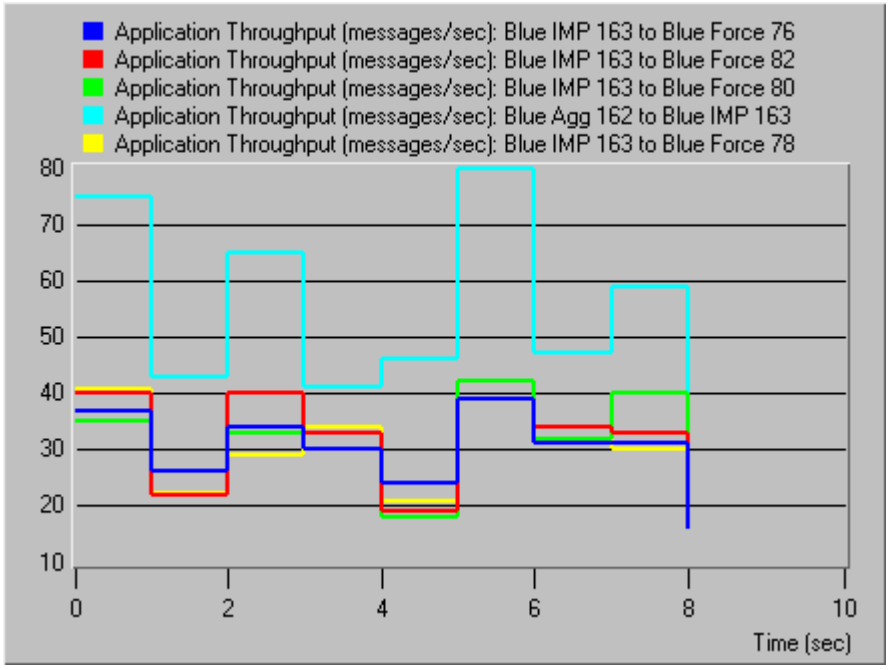


Figure A-4-15: Application Message Throughput Integrated View – Blue IMP 163

Also notice that the general message pattern behavior is representative of an ON/OFF type traffic source and also appears to have uniform ON/OFF periods. This behavior pattern is consistent with other simulation experiment observations.

Figure A-4-16 displays the overall flow of application-layer data between tiers focusing on the time interval 5.1 seconds to 5.5 seconds. Notice that the arrival of a message at the Blue IMP 163 from the higher layer tier, the Blue Agg 162, typically spawns a multiple set of messages to the lower tiers. In some cases, there is a significant

time lag as to when this occurs. The dominant delay measurement is attributed to this tier processing at the Blue IMP 163.

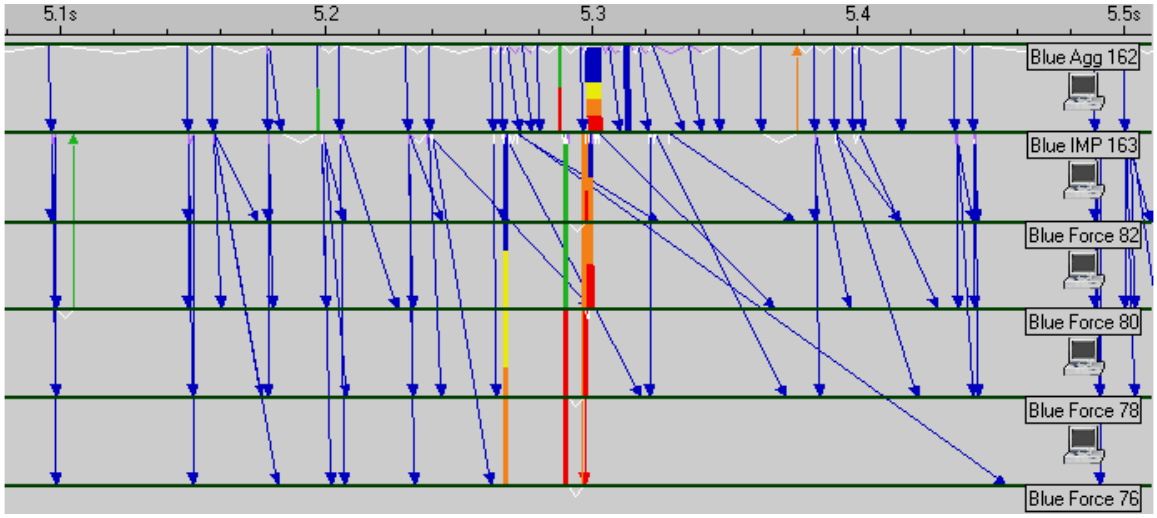


Figure A-4-16: Tier View of Message Flow

All message exchange in the distributed simulation used TCP with a typical message size of 1460 bytes. In analysis of the supporting data, this size was controlled by MTU size settings in the application. Therefore, a typical information exchange at the application level resulted in multiple messages being transmitted in groups of 5 to 8 messages. There were few instances observed where this was not the case. This seemed to result from the message traffic being dominated by the background geography source and the associated periodic update process. The background was simulation of an urban environment with real time objects being simulated such public transportation systems, automobiles and pedestrians. Because the application generated periodic updates to this environment, the result was a message flow behavior that could be characterized as

ON/OFF with uniform rate of transmission while ON, but varying lengths of time of being ON.

Because the application aggregated update information into lengthy messages, it was not possible to determine the flow rate directly relative to an individual element or object in the simulation. To estimate the flow rate, it is required to know the average number of active elements in the simulation. This information was not available. The other factor that is important to acknowledge in this experiment is that the message flow statistics includes ACK messages generated by the TCP. The impact of this is observed by the fact that the average network packet size was in the range 1200 bytes, significantly less than the average 1460 MTU size in observed in the TCP payloads for the message. This indicates an estimated throughput efficiency of 82 % relative to using UDP.

VI Summary of Observations for Message Flow Characterization

Table A-4-2 below summarizes and compares the three message load studies. Two of the three samples reflect an ON/OFF traffic pattern. These two, the Ops simulation and the JFCOM experiment, represent more realistic scenarios than the WSIM RTI experiment as they were very interactive and involved many Federates that had different characteristics of movement.

The WSIM RTI experiment represented a sea vessel moving in a continuous circle and therefore resulted in a strictly uniform stream of update messages. While this simulation is very simple, it does represent what might be expected from any simulation that would contain a continuously moving Federate or any Federate that changed state

from being stable in a visual space to one of moving. This example could be described as continuous flows of messages.

These experiments represent important observations and provide evidence to support our approach for generating message flows for performance measurements and also support the basis for our proposed analytical model presented in the next section. They also provide evidence to support a protocol that manages streaming type message flow.

Table A-4-2: Message Flow Comparison

Characteristic	WSIM RTI	Ops Simulation	JFCOM Experiment
Network	Wide area Internet	Host network hardware interface	Wide area Internet
Transport Protocol	UDP with multicast service via XOM	UDP with broadcast to network interface	TCP
Simulation Object average generation rate (messages/sec)	4	3.0	Not directly measurable as the application aggregated updates. The aggregation process was not visible to the data capture process.
Average application message size (Bytes)	Federate – 170	145	8.7k – 15.7k
	WSIM – 1400		De-aggregation message size was 1448
Average Network packet size (Bytes)	Federate – 212	187	1514
	WSIM – 1442		
Message arrival time distribution	Federate: Deterministic at 4 m/sec	ON/OFF with period random ON – 100 m/sec OFF – 80 m/sec	Federate: ON/OFF with period uniform ON – 40 m/sec OFF – 20 m/sec
	WSIM: Primarily message flow of 8 m/sec and with periods of minor deviations to 6 and 9 m/sec.		Aggregation Point: ON/OFF ON – 80 m/sec Off – 40 m/sec

SECTION 5 MESSAGE LOAD ANALYTICAL MODEL

I Introduction

A major complexity in developing initiatives for distributed real-time simulations is being able to understand traffic loads offered to an open network in order to predict what expected performance might be achieved. It is important to have a sound analytical model to form the basis for being able to describe traffic characteristics with appropriate metrics that can be measured in real application environments. These metrics can then be used with the analytical model to help predicate expected performance and provide a measure of scalability across an open network. These metrics take into account packet error and loss rates, throughput, latency and jitter, and path flows including blocking.

Unfortunately, few controls exist at the host or application layer that can directly influence each of these metrics at the network layer particularly where the application is real-time such as that represented by real-time distributed simulations. These applications, however, can control sending rate and message size. Also, the applications typically have the ability to prioritize messages. It is therefore desirable to have an analytical model that has a basis in metrics that are feasible to measure in the operational overlay network that provide a relationship in a statistical sense to those at the network layer. This allows for implementation of controls and routing algorithms that can be based on these performance metrics.

Part II of this section presents an initial model developed to support understanding performance of these applications. It is analytically based on the sum of exponentials and related assumptions derived from discussions with simulation experts and visual observation of object behavior in a simulation. This model was implemented in C++ code as a message generator used in testing of SRMP and early testing of the XOM prototype.

Part III presents the improved analytical model that allows for use of directly measured performance indicators available in the operational overlay. This approach was fostered by the knowledge gained through the live simulations studies described in the previous section and a better understanding of what is achievable in a overlay network system. The model is based on an ON/OFF traffic model that has derivation from similar models developed for integration of voice and data and Asynchronous Transfer Mode (ATM) communications [Kucz73]. Included is a discussion of a simulation model written for validation of the Poisson arrival process assumption used to simplify the analytical model approach for aggregation of the ON/OFF traffic loading.

II Exponential Traffic Load Model

The previous section provided results of studies of three simulation environments that represent the complex nature of the problem It is complicated by the nature of the application environment as the traffic load is not only specific to a particular simulation, but also to a simulation scenario. To facilitate early studies of traffic load behavior of protocols supporting the RT-DVS, a traffic model was developed for use in the GMU C3I Center Networking and Simulation laboratory (NETLAB) [Wint87]. The development of

the model resulted from application of a general systems engineering approach where a number of informal discussions with experts and observation of visual simulations where conducted. The traffic was first used to aid performance evaluation of the Selectively Reliable Multicast protocol developed at GMU C3I Center [Moen01].

The model was constructed around a 4-state process to represent status of a simulation object in a visual space, where the states represent the status of an object in a distributed real time simulator as presented in Figure A-5-1. The model was developed to include the two classes of messages defined in SRMP: Mode 0 message implies no reliability required and a Mode 1 message requires reliability. The four states are:

1. Object *Off* state: In this state the object is "off" and is not generating any messages.
2. Object *Idle* state: The simulation object is active in the visual space of the simulation, but is not moving and is stable in a multicast group. In this state, the object will generate Mode 0 packets at fixed rate. The object will infrequently (exponential distribution) generate Mode 1 messages.
3. Object *Moving* state: The simulation object is moving within the current multicast group. In this state the object will generate Mode 0 packets at a fixed rate. The object will infrequently (exponential distribution) generate Mode 1 messages.
4. Object *Join* state: The simulation object joins a new multicast group. In this state, the object will generate Mode 1 messages at fixed rate until established in the new multicast group.

- OFF: No messages
- IDLE: Object stable in a multicast group
- MOVE: Object moving within multicast group
- JOIN: Object joins a new multicast group

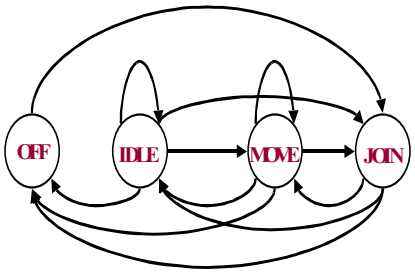


Figure A-5-1: State Transition Diagram

The model was derived simply by ascribing expected behavior by expert knowledge of what objects in a live simulation might be expected to do. For example, the object may not be participating, i.e. the OFF state, or it may be an active participant but the level of activity is such that few update messages are generated such as a simulated vehicle that is stopped. In other cases, the object may be moving in the visual space in which case, more frequent status messages are generated. The JOIN state is used to represent a need for reliable message exchange such as the object join or leaving the simulation or simulation group.

A set of rules where developed that describe this object behavior and formed the logic of the message generator. Without any actual knowledge of live simulation data, exponential random variables where used to drive the logic rules to generate messages assumed to be representative of actual simulation object's message generation process. The result of this approach is an analytical model that is best described as the sum of exponentials.

This model represents a discrete event continuous-time Markov chain [Tijm94]. The state transitions are governed by a Markov chain in which the dependency of the successive state of the simulation object is only dependent on the object's current state. The holding times in each state are, however, exponentially distributed, therefore defining a continuous-time Markov chain.

The message generator model is defined as a continuous-time stochastic process $\{X(t), t \geq 0\}$ with discrete state space I and

$$P\{X(t_n)=i_n|X(t_0)=i_0,\ldots,X(t_{n-1})=i_{n-1}\} = P\{X(t_n)=i_n|X(t_{n-1})=i_{n-1}\}$$

$$\text{for all } 0 \leq t_0 < \ldots < t_{n-1} < t_n \text{ and } i_0, \ldots, i_{n-1}, i_n \in I.$$

Under this definition, two rules are established for the discrete state space to change state:

1. If the system jumps to state i, it stays in state i an exponentially distributed time with a mean μ independently of how the system reached state i and how long it took to get there.
2. If the system leaves state i, it jumps to state j with probability $p_{ij}(j \neq i)$ independent of the duration of the stay in state i, where $\sum_{j \neq i} p_{ij} = 1$ for all $i \in I$.

Given this definition of the model, a traffic generator was programmed in C++ to represent this four-state model. Messages are generated and transmitted to the multicast group at a rate and type (Mode) for each object based on the current state of the object. In the current characterization of the model, all distributions are assumed to be exponential; though the model is built to allow substitution of different distributions for special case analysis or where well known distributions exist that prove to represent distributed virtual simulation data better than the exponential.

While the ability to write a program to simulate or, in this case, to generate message traffic, is relatively straightforward, it is not as easy to represent this sum of exponential distributions analytically. The algebra becomes quite extensive if more than two random variables are involved, though techniques exist to do this. It is also possible to calculate the expected value of the combined random process, though not computationally easy. This aggregate expected value then can be related to actual measurement of average load in the overlay network, though it is not necessarily possible to relate this to individual parameters in the underlying individual random processes.

To develop the aggregation of flow, an Open Jackson Network is used [Gros98]. In an Open Jackson Network, messages can arrive from outside any node and represent a distributed simulator(s) at the node generating packets according to a Poisson process. The mean external arrival rate to node *i* is represented as γ_i, and λ_i represents the total mean flow rate into node *i*. The following set of equations result from the need to satisfy flow balance at each node:

$$\lambda_i = \gamma_i + \sum_{j=1}^{k} \lambda_i * r_{ji} \qquad (0\text{-}1)$$

This is illustrated in Figure A-5-2.

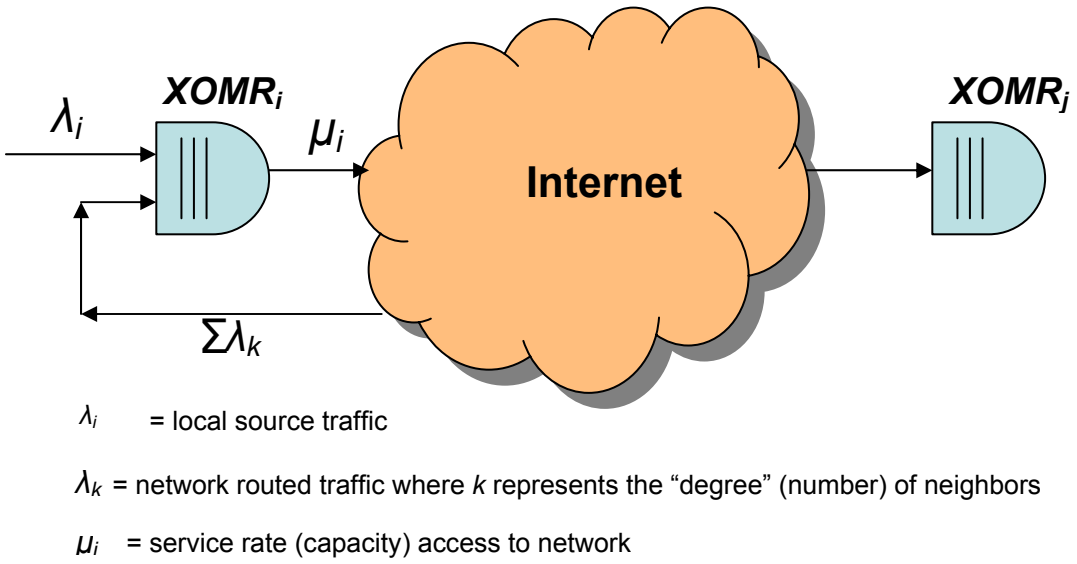


Figure A-5-2: Node Queue Model of Arrival Rates

All servers at node i work according to a deterministic distribution with service rate μ_i. When packets complete service at node i, they go to node j with probability r_{ij}. In this model there is no limit on queue capacity at a node, though an aggregate arrival rate greater than a service rate is not allowed. Each node is then viewed as an independent M/D/c queue with parameters λ_i and μ_i.

The flow balance equations are now solved for each of the types (Mode 0, 1) of messages. The next step is to solve for the number of messages in the queue at each node, L_i using a M/D/c model [Gros98]. The average waiting time at each node is then obtained from Little's formula, $L_i = \lambda_i * W_i$, and assuming that all messages have the same average waiting time, since they have identical service-time distributions and wait in the same first-come, first-served queue. The average system size for a message type can be obtained by weighting the node average total size by the message types t relative flow

rate. Then equation (0-1) is solved independently for each class $\lambda_i^{(t)}$ and results added, $\lambda_i = \sum_{t=1}^{n} \lambda_i^{(t)}$. Then

$$L_i^{(t)} = \left(\frac{\lambda_i^{(t)}}{\lambda_i^{(1)} + \lambda_i^{(2)} + \mathrm{K}\ \lambda_i^{(n)}}\right) * (L_i) \qquad (0\text{-}2)$$

In this way, representing flow in a multicast environment presents a simplification of the numerical analysis of the model. Multicast implies that the traffic flow is replicated at each node in the network. This implies a routing matrix that contains only zeros or ones. The analysis then can proceed from a left to right flow for a given multicast group instance under consideration and is represented by a multi-commodity maximum flow problem. (The approach is not necessarily constrained by a 0/1 routing matrix. The multi-commodity max-flow problem can be solved without this constraint, however, negative flows are not allowed.)

Multi-commodity flow problems [Bert98] are a class of network problems that involve several commodity flows that are couple together either by cost or flow bounds. The model uses flow bounds in the form of bandwidth constraints. This approach allows solutions for separate commodities and source-destination node pairs. Left to right flow instances allows simplification of the network to a minimum spanning tree without any loss of analytical interest. After reduction of the network to a minimum spanning tree, analysis proceeds with solving the traffic flow equations and the queueing problem defined earlier.

For our example, Figure A-5-3 represents a seven node, seven link network with external traffic input at each node.

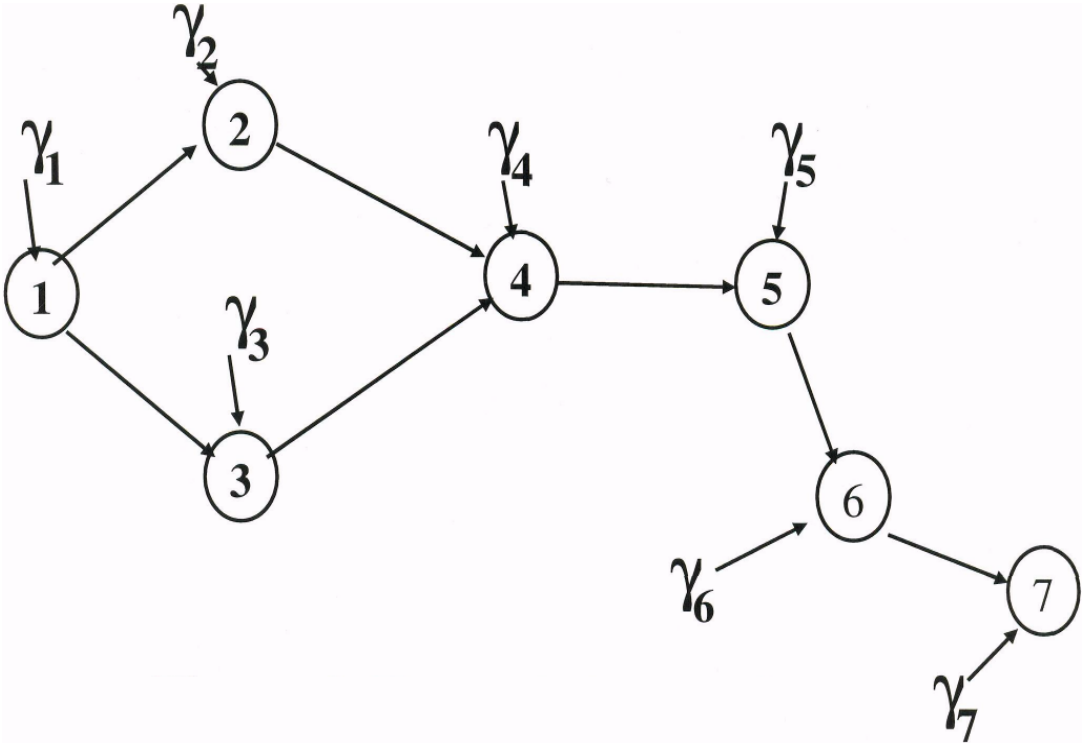


Figure A-5-3: Example Network

The minimum spanning tree reduction, presented in Figure A-5-4, reduces the network to a seven node, six link network. The reduction is allowed since we are interested in the flow into node 7; traffic originating from node 1 destined to node 2, leaves the network at node 2 and is not forwarded. (While node 2 performance may be of interest, performance analysis of node 2 can be solved as another instance of the problem where the minimum spanning tree is two nodes, with one link.) The same node 1 traffic is replicated and offered to node 3 in the minimum spanning tree.

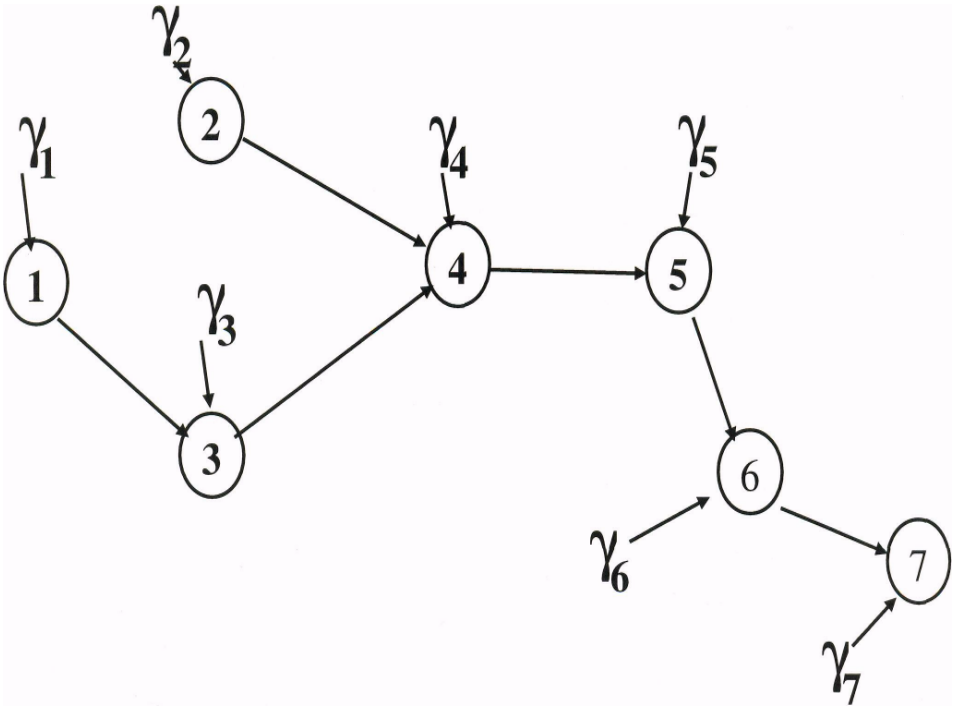


Figure A-5-4: Minimum Spanning

The problem can be set up to analyze performance at any of the nodes by adjusting the routing matrix in the Jackson network analysis, e.g. make another node a sink node, define the resulting minimum spanning tree and perform the multi-commodity queueing analysis.

This analytical approach proved useful for writing a program to generate message traffic in the absence of any characterization knowledge of the RT-DVS environment; however, it is not easy to understand the performance in an analytical sense using this approach. The complexity of the algebra makes supporting dynamic or large network environments with it infeasible. In addition, the analytical approach implies that there are no restrictions on queue capacities at each of the nodes in the network and state-

independent routing probabilities can be applied at each node in the network. This is not the case for RT-DVS as is demonstrated by the results of prototype performance testing presented in Section 6. The knowledge gained from studying this early analytical strategy resulted in a much improved model that can readily be applied to the overlay concept. This model is presented in the following paragraphs.

III ON/OFF Traffic Model

The motivation for a good and efficient analytical model is to reduce dependencies on simulation to provide the basis of understanding network performance of the overlay. In addition, it is highly desirable to have a near real time correlation of measured traffic load to a simple analytical model so that a routing algorithm can use the information for dynamic optimization of the overlay network. This motivation is not new and has fostered many ideas to accomplish this. Heffes [Heff86] introduced the idea of Markov modulated characterization to support statistical multiplexing of packetized voice sources together with data traffic using the Markov modulated Poisson process (MMPP). The approach is used to support ATM networking analysis and design. Kang [Kang95] refined the approach for two-state MMPP modeling of superposed traffic streams.

Ma [MaJo04] expands upon these ideas and presents a simple, physical-based traffic model using a two state (ON/OFF) Markov process to model traffic from heterogeneous sources. The approach results in ability to express traffic statistics in terms of three physically identifiable or measurable usage parameters: bandwidth used when ON, fraction of time utilized, and the number of ON times per unit of time.

The traffic characterization studies of real time simulations presented in Section 4 showed these traffic traces to have very similar properties to this ON/OFF process. This ON/OFF approach is analytically appealing as it can be represented by Markov Modulated Poisson Process. It also allows for consideration of other traffic distributions such as Pareto, though analytically, the available statistics change. Specifically, traditional statistics such as mean flow rate or variance are no longer available, as Pareto distributions do not necessarily have all their moments. However, the statistics that are available may still be useful. For example, the probability of an access link being in overload condition and the probability of how long it is likely to remain in overload, statistically, are indicators of stability in the overlay network.

Another reason that this strategy is appealing is that Simon [Simo04] has demonstrated the value of using flow control mechanisms based on this approach. He has presented simulation results of using rate based admission control at the access point to provide a certain level of expected performance across the overlay network. This mechanism provides a measure of control and predicts message loss across the overlay network using a traffic model similar to that presented by Bianchi [Bian00].

Using this ON/OFF strategy allows for representing an aggregated traffic trace from N independent ON/OFF sources. This approach is very representative of the original 4 state model developed for use as a traffic generator described in the previous section. The following paragraphs describe an analytical approach suitable for representation of the expected traffic load that could be used as part of the path management in an overlay network.

According to Fischer [Fisc93], the first use of the MMPP for traffic analysis was in Naor [Naor71]. Many papers and books now provide examples [Kucz73] of the use of MMPP with Fischer [Fisc93] presenting a number of results for easy understanding and establishing the basis for a variety of applications. In the case of our studies of simulation traffic, the characterization reflects the ON/OFF model, however with what appears to be somewhat discrete ON/OFF periods and also where the ON/OFF relationship represents a higher or lower rate of message arrivals rather than on or off. This actually serves to simplify the problem and readily supports our desires to have measurable parameters that can be used to impact the overlay routing performance in line with the Simon [Simo04] approach for admission rate control.

Bianchi [Bian00] provides a flexible approach to representing ON/OFF traffic models. His approach allows for the superposition of independent homogeneous ON/OFF variable rate traffic sources with exponential distribution of ON/OFF times. By assuming independence of each traffic source, the approach allows for calculation of the average source traffic rate by simply scaling by the time ON relative to the sum of time ON and the time OFF.

The independence assumption also allows for defining a process where the probability of i sources over k active at time t as a Bernoulli distribution. This is similar to the discrete time slot analysis for ATM communications system where k represents the time slots and i represents the arrival of packet. This is a discrete time, geometric inter-arrival process which is a Bernoulli process [Pitt00, Star94].

This provides for a straightforward analytical solution to develop the conditional probabilities that a source is in the ON state. However, the next steps in the process model for throughput analysis are complex as the real interest is in having an expression for the capacity received by a new source in terms of capacity already allocated in the access link for a node in the overlay (Simon [Simo04] approach to admission control).

Bianchi accomplishes this by developing an approximation based on a fixed rate traffic model rather than a variable rate model. The entire derivation is available in [Bian00] and the specific results of interest are presented here.

$m(t)$ is defined as the number of additional new traffic sources and $k(t)$ as the number of sources already in service. The assumption is made that new sources fairly share the capacity left available by those already in service, then the rate (or bandwidth) $B_r(t)$ instantly available to the new sources is approximated by

$$B_r(t) = \min\left(B_p, \frac{\max\left(C - k(t) * B, 0\right)}{m(t)+1} \right) \tag{0-3}$$

where B is the traffic rate (bandwidth) of sources already in the system and B_p is the rate of new sources. C is the link capacity of the system.

There are certain properties of this model that make it appealing to the overlay problem. Specifically, we know what the link capacity is or can at least estimate it. We are able to measure or count how many sources are active, at least in a statistical sense, therefore allowing decision making at the local node as to ability to allow new sources into the system without degradation of in service sources.

However, there are two shortfalls in this approach. The first is that the model assumes there are no messages generated in the OFF state. We know from studying the targeted simulation environment that this is not the case. The model would therefore need to be adjusted to reflect this requirement. The second shortfall is that it still is somewhat computationally intensive. This results from the input Poisson process assumption that gives a 2^N state space for N independent sources.

Another approach is to consider taking advantage of traffic studies that indicate that the simulation environment has relatively deterministic traffic generation rates and that the ON/OFF periods generally have relatively uniform distributions of alternating ON and OFF periods. The other observation of the measured traffic is that the simulations tend to have fixed length messages sizes. This leads to the consideration of an approach that uses an aggregated flow M/D/1 queue model which appears to be easier to use and has basis similar to the M/D/c model presented earlier in this section.

For the analysis, an assumption is that application data sources have stationary statistics and are independent. The previous section provided evidence that the message traffic behavior is deterministic, so the assumption is valid. The assumption allows us to then use a multiplier, N, for summing of multiple sources. This also implies that as the number of sources increases, that the traffic load variance will also increase linearly. Ma [MaJo04] and Pitts [Pitt00] use this to simplify the analytical model.

The approach results with the idea of flow aggregation. Pitts [Pitt00] presents an argument that flow aggregation is possible based on the assumption that the output port, or in our case the access link to the network from the XOM, has much greater capacity

than the offered source rate of message arrivals. In the cases studied this assumption also was valid. Further, this assumption can be forced always to be valid in the implementation of the XOM by imposing it as a constraint in allowing new flows or sources of traffic either directly or by using a threshold measurement value that provides warning that the XOM is no longer able to offer expected level of service.

If this threshold is defined as the available service rate *C*, then the rational of Ma [MaJo04] and Pitts [Pitt00] can be used to construct a two-state overflow model where the aggregate process is either in the ON state, meaning the message arrival rate exceeds the output service capacity *C* (or defined threshold) and an OFF state where the arrival rate is less than then the service capacity (threshold), but not necessarily zero.

A measure in the XOMR prototype counts the message arrivals in time intervals and provides for a threshold parameter associated with the capacity of the XOM node. This is implemented using a Web service interface and provides for display of these measures near real time. The two-state approach, therefore, can be used to directly apply performance management to the overlay network in near real time, which is the objective.

The ON/OFF model analytical development details are presented in Kang [Kang95], Bianchi [Bian00], Ma [MaJo04] and Pitts [Pitt00]. An approach customized to this application is presented here for completeness and includes results from a simulation written to demonstrate the ON/OFF traffic relationship to studied simulations and confirms that the assumptions used for the simplification are valid.

The messages generated by a simulation can be characterized as message flow having a mean ON time of T_{ON} and a message arrival rate of λ (messages/s). This gives a per

flow average of $T_{ON} * \lambda$. In the XOM, it is possible to measure the mean load of messages arriving which we define as A_M, then the rate of flows arriving is easily calculated by

$$F_A = \frac{A_M}{T_{ON} * \lambda} \tag{0-4}$$

This traffic model applies to *N* sources of traffic so this must be modified to represent the aggregate which reduces the state space from N^2 to two. This is illustrated in Figure A-5-5. We then define T_{ON} to be the time that the process is in excess of the threshold *C* and T_{OFF} to be the period of time that the arrival process is less than the threshold. The characterization of the RT-DVS presented in Section 4 indicated a deterministic source message size that arrives at a constant rate. This allows describing the service rate as deterministic. In addition, we assume that the overlay host has a single connection to the supporting underlying network. By definition, this allows application of queuing analysis using the M/D/1 queue model for the study of the arrival process and the departure process at an overlay node. Knowing *C* and λ, the maximum number of message flows is

$$N_0 = \frac{C}{\lambda} \tag{0-5}$$

Similar to Pitts [Pitt00], we make the assumption that message flow arrival is memoryless as a result of user activity in the application triggering the flow. Using(0-4), we can define the offered message load in terms of message flows as

$$A = \frac{A_M}{\lambda} = F_A * T_{ON} \tag{0-6}$$

For our analysis, we are interested in the probability of a message being lost at a node as a result of finding the service buffer full upon arrival. Since the application environment is real-time, when a message is lost it is not recovered and leaves the system. This is equivalent to the probability of the message being blocked, that is, the steady state probability that the buffer is full when the message flow arrives. By definition, this allows use to apply the Erlang B call waiting formula described in Gallager [Gall83] and Gross [Gros98]. This leads directly to use of the Erlang call waiting formula as a representation for probability of a message flow being blocked as developed by Pitts [Pitt00] for flow analysis:

$$D = \frac{\frac{A^{N_0}}{N_0!} * \left(\frac{N_0}{N_0 - A} \right)}{\left(\sum_{r=0}^{N_0} \frac{A^r}{r!} + \frac{A^{N_0}}{N_0!} * \left(\frac{N_0}{N_0 - A} \right) \right)} \qquad (0\text{-}7)$$

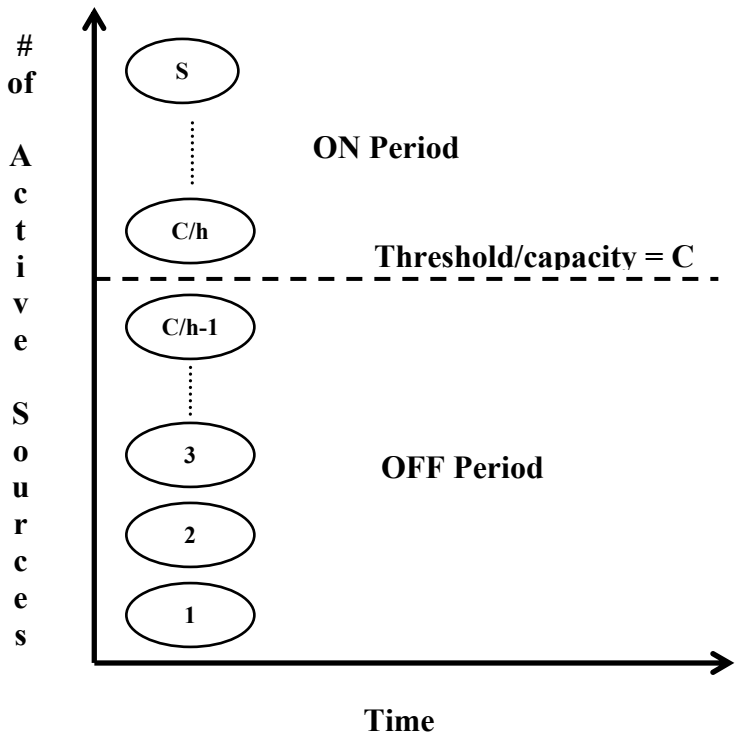


Figure A-5-5: Two State Model for XOM Threshold Capacity

The average number of message flows waiting averaged over all flows is given by

$$w_a = D * \frac{A}{N_0 - A} \tag{0-8}$$

Dividing equation (0-8) by D gives the mean number of flows waiting conditioned on there being some flows waiting. When the aggregate traffic is in the ON state, then the mean input message flow exceeds the threshold C and this excess rate is the product of the conditional mean number waiting and the message rate of message flow or

$$R_{ON} = C + \lambda * \frac{A}{N_0 - A} = C + \lambda * \frac{A_M}{C - A_M} \tag{0-9}$$

Using Little's formula, the mean duration in the excess-rate ON state is

$$w = F_A * t_w = \frac{A}{T_{ON}} * t_w \tag{0-10}$$

and rearranging and substituting for w results in

$$t_w = \frac{T_{ON}}{A} * w = \frac{T_{ON}}{A} * D * \frac{A}{N_0 - A} \tag{0-11}$$

Then the conditional mean delay in the ON state is given by

$$T(on) = \frac{t_w}{D} = \frac{T_{ON}}{N_0 - A} = \frac{\lambda * T_{ON}}{C - A_M} \tag{0-12}$$

To calculate the rate in the OFF state, R_{OFF}, we make use of the fact that the long run probability that the aggregate process is in the ON state is the same as the probability that a message flow is delayed, or D. This is equal to the long run portion of time in the ON state

$$\frac{T(on)}{T(on) + T(off)} = D \tag{0-13}$$

Therefore,

$$T(off) = T(on) * \frac{1 - D}{D} \tag{0-14}$$

The mean load in messages per second as the weighted sum of the rates in the ON and OFF states is

$$A_M = D * R_{ON} + (1 - D) * R_{OFF} \tag{0-15}$$

Therefore R_{OFF} is

$$R_{OFF} = \frac{A_M - D * R_{ON}}{1 - D} \tag{0-16}$$

This completes the development of an aggregated arrival process to allow describing the process in terms of packet flows. This approach leads directly to the concept of associating these flows with group management in the overlay network. A source would generate packet flows to a multicast group. These groups can then be aggregated for efficient routing in the overlay network. The aggregation of flows can be represented in this analytical model and allow for use in understanding queuing behavior in the XOM as presented in Figure A-5-2 above.

Adopting the assumption that the excess rate message arrivals in the ON state are geometrically distributed and that the free periods, or the periods of less than excess rate arrivals, are also geometrically distributed, then the queue overflow probability is given by Pitts [Pitt00] as

$$P(x) = \frac{\lambda * D}{C - A_M} * \left(\frac{1 - \frac{1}{T_{ON} * (R_{ON} - C)}}{1 - \frac{1}{T_{OFF} * (C - R_{OFF})}} \right)^{x+1} \tag{0-17}$$

The firs part of this expression represents the probability that an arriving packet is an excess-rate arrival given by $\frac{\lambda * D}{C - A_M}$. The second term in the expression represents the geometric progression of the status of the queue and is the probability that the queue exceeds *x* messages. Since this are independent events, multiplying the terms together results with desired expression for queue overflow probability as presented by equation (0-17).

IV Poisson Assumption

Earlier in this Section, a key assumption allowed the straightforward use of the aggregation model based on M/D/1 analysis. It is important to validate this assumption using the understanding gained in the real time simulations studied. The assumption was that this is a memoryless process therefore allowing a Poisson arrival.

To understand what the impact of using the Poisson arrival might be in the analysis of a process where there are a large number of sources being aggregated, a computer program was written to simulate the generation of a large number of ON/OFF sources where the message arrival process used a random exponential distribution for source generation. Each source generates messages at the rate of $\lambda = 4$ messages per second when in the ON state. This rate was chosen as representative of the observed rate in the traffic studies. The ON/OFF times of the sources were equally weighted as this was also the observed case in the studies of live simulations. The program was written to generate 10^6 messages. Accounting was accomplished by noting message arrival time in simulated time. The program first counts and records the number of messages generated in 10ms interval and then further aggregating the counting process for 100ms and then 1000ms. By calculating the mean and variance of each of this counting process, it allows for testing the similarity of the message aggregation process. The details of this procedure are described by Stallings [Stal98] and Embrechts [Embr02], who define the parameter $\beta\left(0 < \beta < 1\right)$ to indicate the degree of self-similarity. β is then used to calculate the

Hurst parameter [Stal98] as $H = (1 - \frac{\beta}{2})$. An $H = 1$ would indicate a high value of self-similarity in the aggregate process and $H = .5$ would indicate no self-similarity.

The objective here is to validate that our Poison assumption is valid, a known stationary process. If the process is at least close to stationary for a large number of sources, then our approach for aggregation seems likely to be valid for our intended use. The results of the simulation for 100 sources are presented in Table A-5-3. Notice that the average rate of message generation does not appear to change across the counting process; it only scales relative to the order of magnitude change of the counting process. This implies that the aggregation does retain some of the original characteristics.

Table A-5-3 Computer Simulation Results of Traffic Aggregation

Counting Interval	Mean	Scaled Variance
10msec	3.247986	0.012168
100msec	32.4791531	0.0497741
1000msec	324.791531	0.58878
10000msec	3247.9667	5.60284

It is also necessary to test the scaled variance, by plotting the logarithm of the time scaled (normalized) variance against the logarithm of the counting instance. The results of that are presented in Figure A-5-6. A straight line is fitted to the data points with the slope calculated to be 0.906. A slope near one would indicate a stationary process. Therefore, the figure indicates a slight deviation from a completely stationary process.

This may not be unreasonable as the data are obtained by summing a series of exponentially distributed processes; therefore, we might expect a slightly heavy tailed distribution as a result. This result is similar to that described by Erramilli [Erra96] in analyzing long-range dependent packet traffic.

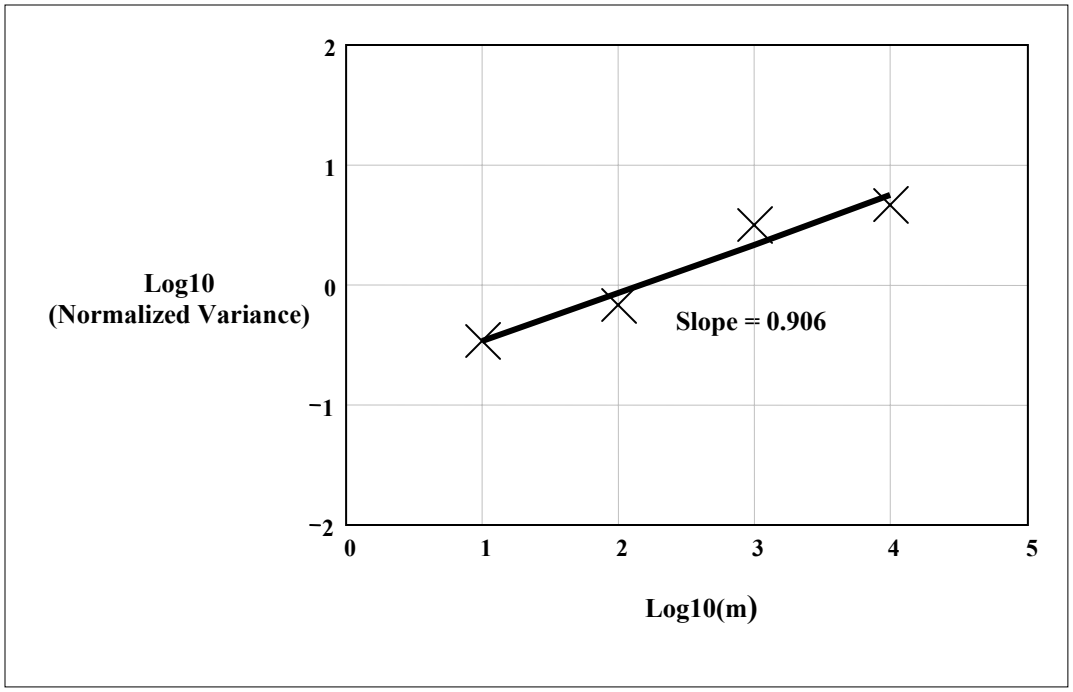


Figure A-5-6: Plot of $\log_{10}\{Var[X^{(m)}]/Var[X_t]\}$ **versus** $\log_{10}(m)$

If we apply the Hurst analysis to our data, the result is $H = (1 - \frac{0.905}{2}) = .5475$ which indicates a slight deviation from a completely ergodic process. We actually desire to have some deviation as the traffic patterns observed in the live simulations indicate a somewhat repeating traffic pattern which indicates a deviation from a pure ergodic

process. The conclusion is that the Poisson assumption is adequate for this ON/OFF model.

V Summary

The Section provides for an analytical approach to form the basis for understanding the overall expected performance of overlay multicast in the distributed simulation environment. The approach focused on having an analytical model that has directly measurable parameters available in the actual operation environment of the overlay. The concept is that these parameters could then be used in a routing algorithm for delivery of messages with consideration of the expected performance. This is important for overlay multicast as the overlay protocol essentially has no ability to control the service performance of the lower layer network.

Part II of this section presented the first model developed for support of understanding performance of these applications. This model proved to be representative of actual traffic and therefore useful in the performance understanding of the prototype protocol in the absence of live traffic. The model was based on the sum of exponentials as derived from observing behavior of a single object in a visual simulation. The model is implemented in C++ and has been used early prototype testing of XOM.

The knowledge gained in this early traffic generator was applied in the development of an analytical model that allows for use of directly measured performance indicators available in the operational overlay. Part III of this section presented the proposed analytical model. The approach was based on an ON/OFF traffic model that has

derivation from similar models developed for voice and Asynchronous Transfer Mode (ATM) communications. Part IV of this section presented results of a simulation model written for validation of the Poisson arrival process assumption used to simplify the analytical model approach for aggregation of the ON/OFF traffic loading. The next Section demonstrates the use of this analytical model in some examples that are specific to the XOMR and relate to measured performance results of the laboratory test of the current XOMR prototype.

SECTION 6 PERFORMANCE CONSIDERATIONS FOR OVERLAY MULTICAST

I Introduction

The previous section presented an analytical approach to understanding the performance of overlay multicast in support of distributed simulations. This section provides a link between the analytical model and evaluation of an early prototype of the XOM overlay. The purpose is to validate the analytical model and serve to form a basis of the proposed architecture.

Part II of this section presents results of laboratory tests of the current XOMR prototype conducted in the GMU C3I Laboratory. The testing objective was to demonstrate message throughput of the basic message routing functions. The specific performance interest in the testing was to measure message throughput of the send/receive functions and relate this to an understanding of the impact of the *n*-degree or the number *n* of reflect messages that might be achievable from an XOMR. The *n*-degree factor is a significant parameter as it represents the ability of the XOMR to replicate messages to multiple paths or channels, therefore a driving force behind overall path construction and path optimization.

The second factor of interest in the XOMR performance is the optimum diameter of the overlay system, or the number of XOMRs that could conceivably be cascaded before message loss would be unacceptable. For example, if the end-to-end message loss

objective across a ten-node overlay system is be to less than 1%, then the loss allocation to each XOM node must be less than 0.1%. Measuring message loss in the laboratory tests and using the information for validation of the analytical model also gives a tool to help establish thresholds that can be used for determining operational overlay system diameter.

The third facto of interest is end-to-end path delay. This is estimated by summing the processing delays of the nodes in the path as well as the link transmission delay across the underlying network between the overlay nodes.

Part III of this section discusses key aspects of the proposed architecture influenced by results from laboratory tests as well as from the study of the live simulations discussed in Section 4.

II Performance Studies of the XOMR Prototype

The initial test environment was established in the George Mason University (GMU) C3I Center Network Modeling and Simulation Laboratory. The test configuration is presented in Figure A-6-1. The configuration consisted of four XOMRs, each operating on separate subnet LAN segments. These four segments were connected together via a single IP router to represent a WAN. The C++ MulticastRouter module was used in the XOMRs, providing better performance than the original Java MulitcastRouter This configuration represents a 3-degree routing function, where degree means number of paths required for the XOMR message replication and forwarding.

A second configuration also was tested, where only two XOMRs where connected across the router. This scenario was design to give a measure of maximum throughput based on a single channel. While this scenario is not likely to be a normal operational scenario, it does provide a baseline throughput test that is used to compare higher degree scenarios. This is important for establishing expectations for performance and benchmarking for XOMR code optimization in future releases.

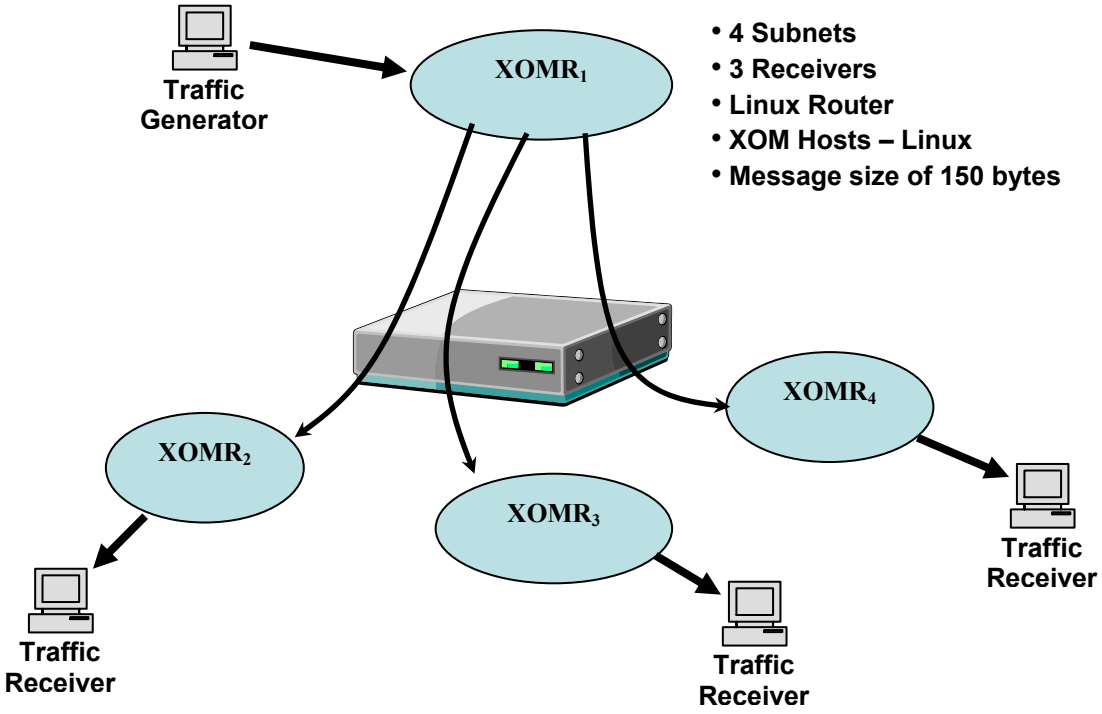


Figure A-6-1: Laboratory Test Scenario

The applied message load was increased gradually in a series of test runs with the performance measure being lost messages at each load measurement point. The loss ratio

was calculated in plotted against offered message load. Multiple runs where conducted. The summary results are presented in Figure A-6-2. Loading tests where stopped when traffic loads caused the loss ratio to exceed 1 %. This point represents the design performance goal of a single XOM and assists in establishment of an operating threshold. This threshold is then used to establish an operating point so that a single XOM node does not have message loss greater than 0.1%. This maintains the integrity of the overlay system under the definition of overlay diameter described earlier.

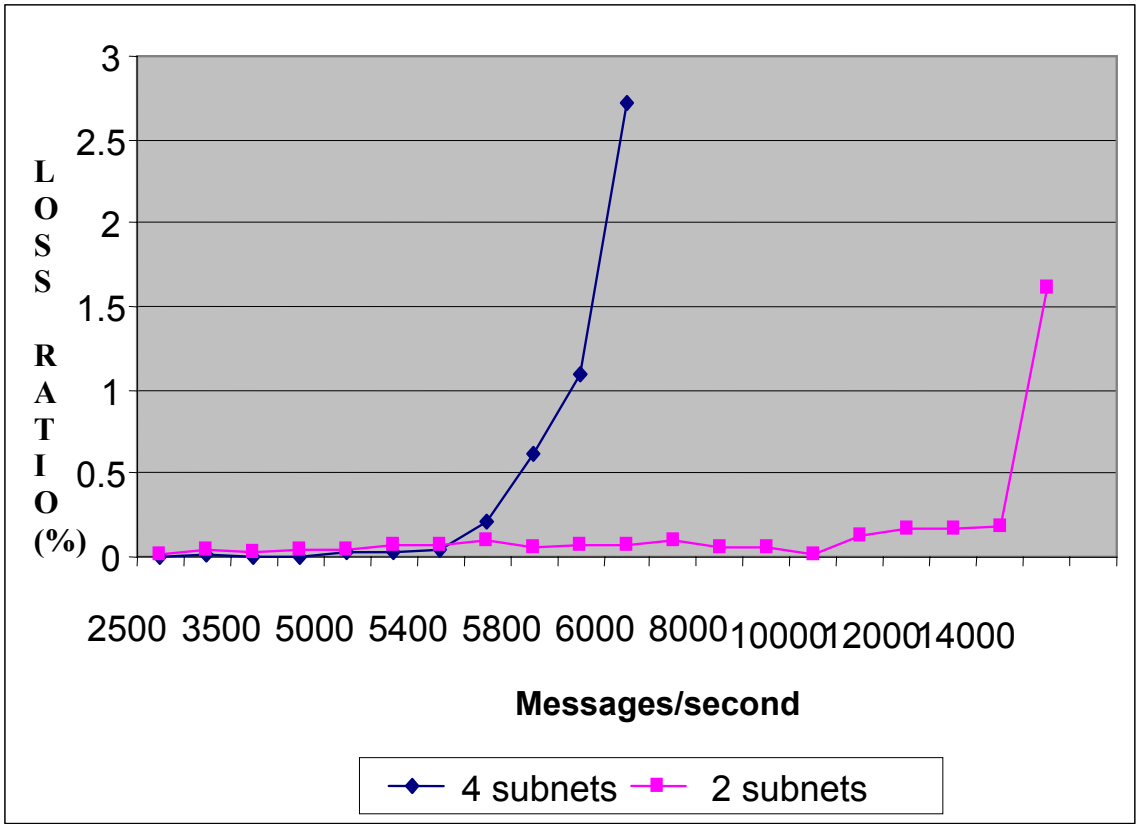


Figure A-6-2: XOMR Loss Ratio (%) Performance Test

The XOMR message throughput was approximately 5900 messages per second for the 4-subnet scenario and nearly 15,000 messages per second for the 2-subnet scenario. There is approximately 60% reduction in throughput going from the 2-subnet case to the 4-subnet case.

To understand this greatly reduced behavior in performance with the 4-subnet scenario, the detailed timing diagram of Figure A-6-3 was captured using OPNET ACE. Two messages were forwarded as quickly as possible in sequence to the XOM. This allowed for analysis of queue behavior of the XOM when a message is in service and

another arrives. The analysis of the timing relationship between arrival of a message at the XOM from the generator and the subsequent replication of the message by three provides some understanding as to why there is a significant reduction in message throughput for the multiple replication case.

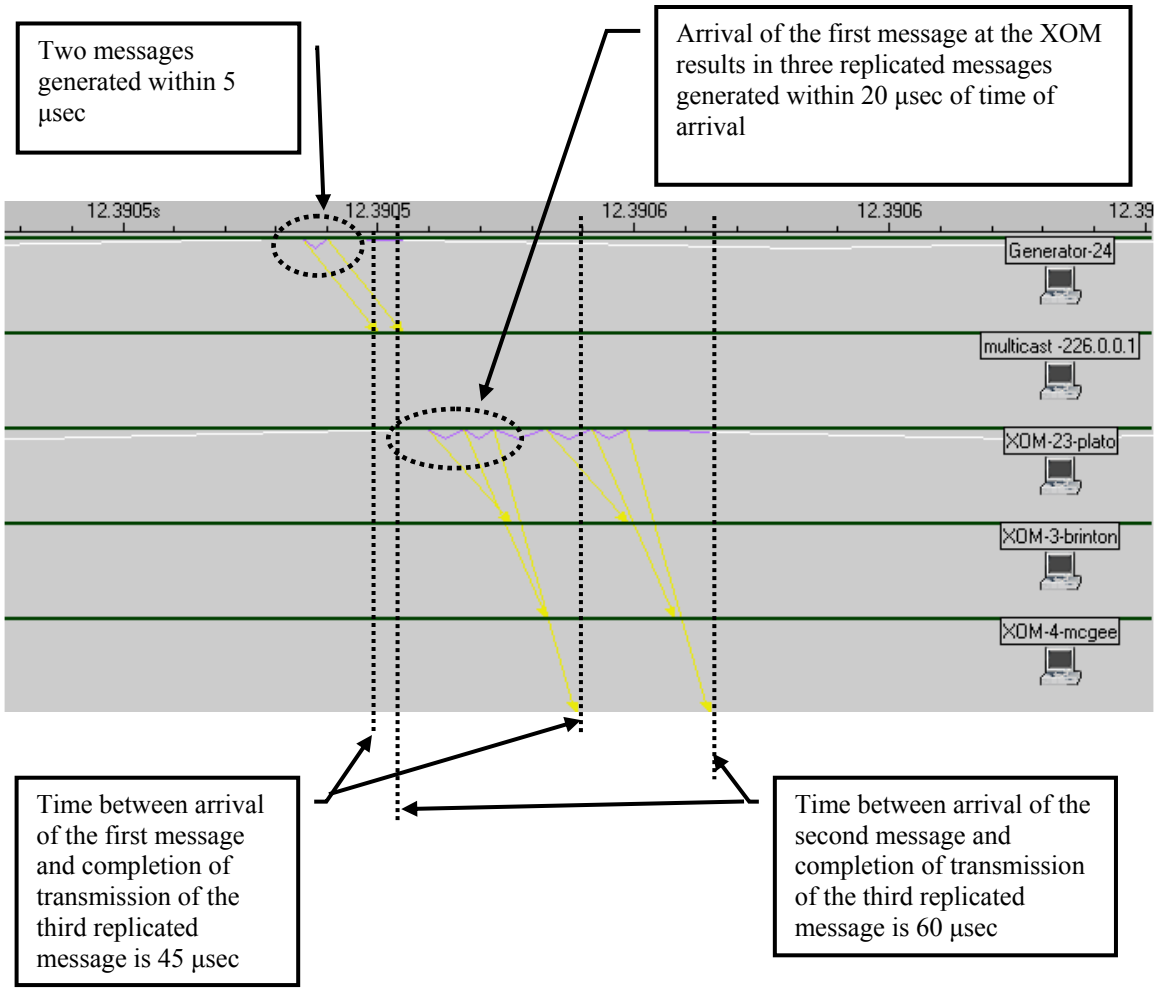


Figure A-6-3: Tier Processing Timing of Message Arrival

The graph indicates the impact of an arrival of an additional message when one message is already is in service. When the first message arrives, it enters into service and completes the replication of three messages before it can start processing the arrival of the second message. Since the second message must wait for the first to finish service, it appears to take more than twice the time required by the first before it completes processing even though it arrived only 5μsec later than the first.

Considering the total process including the time for completion of service (processing of the arrival and replication and transmission of three messages), then the analysis of a single arrival implies that the service rate expected should be $1/.000045 = 22.2$k messages per second. For the second message, the implied service rate is $1/0.000060 = 16.7$k messages per second.

If the arrival process was a memoryless one such as Poisson, then queue build up and overflow occur in a gradual process. The expectation for such an arrival process is that the system would respond by servicing the queue in the gaps of new random arrivals which tends to smooth out the knee of the failure curve or have a more gradual failure characteristic.

If the system is characterized by more deterministic arrivals (e.g., the arrival process has memory), then the result has a sharper curve of when the queue overflows. This behavior is similar to that observed in Figure A-6-2 and results from the fact that the message generator used was deterministic with fixed length message sizes. This characteristic is similar to that observed in the live simulations presented earlier.

The first laboratory tests conducted considered batch arrivals similar to the situation presented in Figure A-6-3. The results of these tests had throughput performances in the range of 2,000 messages per second, which is significantly different than the performance that appears possible based on the single observation in Figure A-6-3. This observation combined with the results of applying a more deterministic distribution, imply that the prototype performance can be improved.

To help understand the throughput from a queue wait time perspective, we can apply the M/D/1 queue model developed in the previous section. From Figure A-6-3, we can estimate the message processing time for the first message as approximately $35\mu\text{sec}$. Using this estimate, the M/D/1 delay model is used to calculate the curve in Figure A-6-4. The resulting curve gives a range of utilization relative to expected queueing delay. This curve provides operational guidance on the level of utilization that should be targeted for the XOM and the impact on expected system delay based on current prototype performance. In the test case presented here, targeting a utilization threshold of .7 implies queuing delay of approximately $40\mu\text{sec}$.

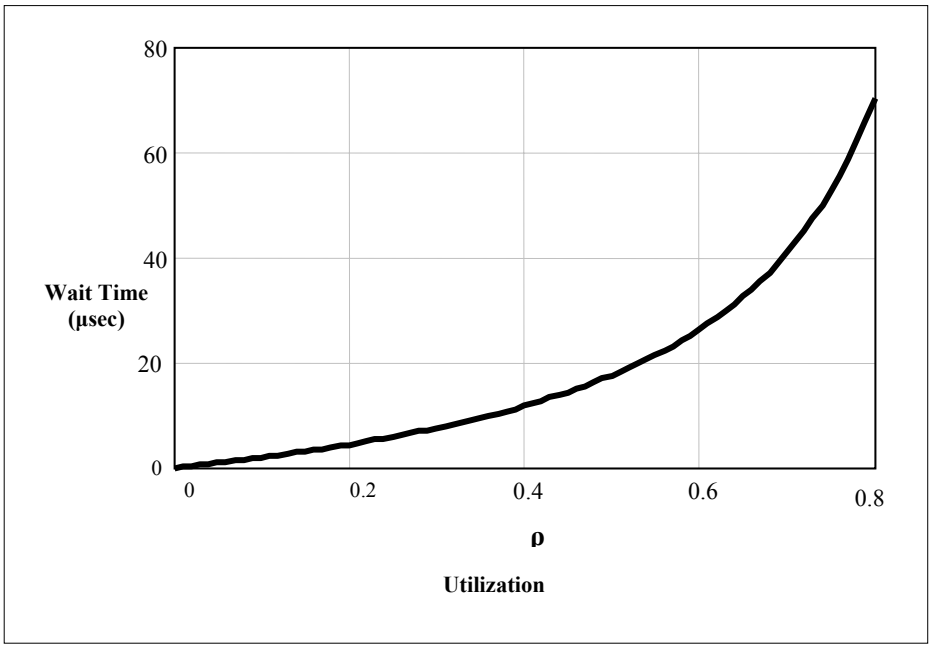


Figure A-6-4: M/D/1 Queue Waiting Time for Message Processing Time of 35μsec

Continuing with this model and assuming that the measured 5900 message rate at 1% overall loss is the capacity, then we can formulate the probability of overflow model described in the previous section. If we apply the .7 utilization factor as a desired threshold, and assume this to the measured rate of average load and that the Java interface socket has a queuing capacity of 64,000 bytes (approximately 400 messages), then we get the probability of aggregate overflow as presented in Figure A-6-5. The curve represents expected performance at 80 % of the XOM measured capacity which results in queue overflow probability of 0.1 at buffer size of 400 messages. This curve then predicts performance for the Java interface socket.

This can be also used to aid overlay system design in terms of the optimal diameter of the operational overlay network. For example, if we desire to have an end-to-end system

message loss rate of 1% and each XOMR in the system represents a loss of 0.1 %, then the system diameter based on message loss is 10 nodes.

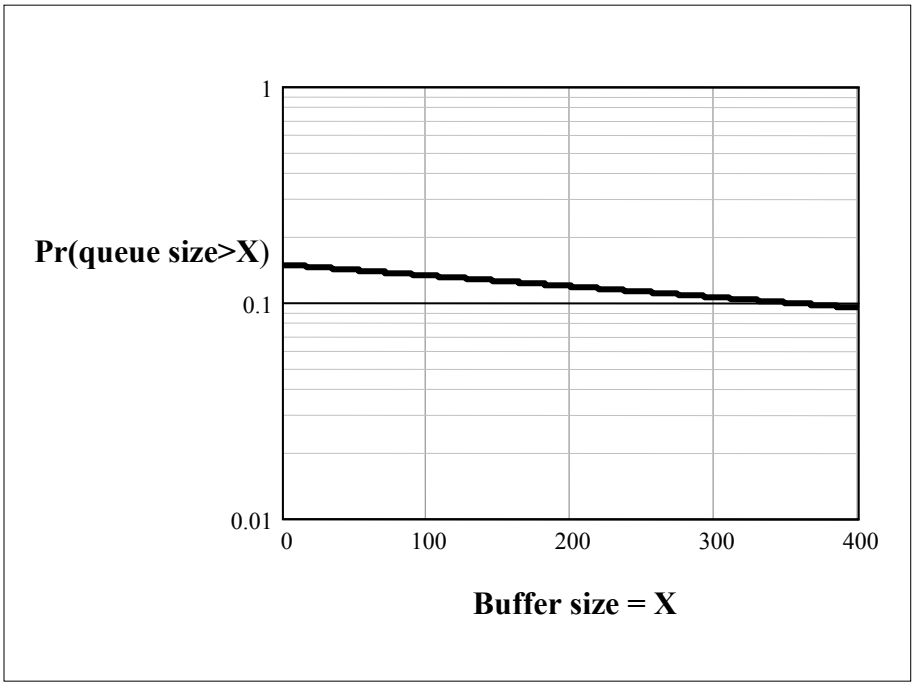


Figure A-6-5: M/D/1 Aggregate Probability Queue Overflow

Using the estimated processing delay of a node in the overlay, we also can define the end-to-end path delay of the system. The end-to-end path delay across the overlay is simply the sum of individual node's processing delays in a path of interest plus the inherent network delay due to distance across the underlying network between the nodes. This can be expressed as

$$D_{\text{path } i,j} = \sum_{n=i}^{j} D_{\text{node}_n} + \sum_{n,m=i,x}^{x-1,j} D_{\text{Link}_{n,m}}$$

where *n,m* represent the end nodes of the links along the path from *i* to *j*. The first summation is the processing delay at each node along the path and the second summation is the link delay between the nodes along the path.

This part of section 6 has presented results of performance testing of the XOM prototype. Included were traffic loading tests using a 4-subnet and a 2 subnet test scenarios. These tests were used to find the point of traffic load where the XOM message loss ratio exceeds 1%. Based on study of a single message arrival and the closely spaced arrival of a second message, measurements were made of the XOM processing time for replication of messages. These measurements were used to estimate what might be possible for throughput performance in an optimal XOM solution. There are two important results from this section. The first is the observation of importance node degree, or the number of message replications required of the XOM for arriving messages, to the overall performance of the XOM. The second is that the measured performance of the XOM can be related to the proposed analytical model of Section 5. In the next part of this section, these results are used to describe key features of the proposed architecture for the optimal XOM.

III Architecture Considerations

There are many important considerations in the overall design of the XOM that result from the study of the live simulations and the laboratory testing of the prototype. In this part of section 6, the most important of these are presented in relationship to the proposed architecture. This section presented results of laboratory testing of the prototype to

analyze throughput performance and establish the relationship to the analytical model developed in the previous section. The results lead directly to design considerations for end-to-end performance of the overlay. In addition, Section 4 provided insight to the behavior of live simulations that also impact the design. The most important of these are that the message flow is represented by ON/OFF traffic pattern where OFF means a lower rate, never zero, and the message sizes tend to be of fixed length. These considerations are important for XOM to support efficient, reliable many-to-many multicast transmissions over existing network protocols used across open networks.

The XOM must provide real-time response and predictable network services in order for the end simulation systems to interact within specific delay bounds. In the overlay concept, the overly protocol host only has control of its own performance and allocation of local resources and does not have an ability to influence the services of the underlying network. In the previous section, we describe what the limits of capacity are in terms of throughput and packet loss. Both of these parameters are measurable locally and reflect the local capacity. The current prototype now includes a statistics module that captures this data and makes it available for viewing using Web services. The approach not only allows viewing of the local host XOMR performance, but a feature allows viewing of other XOMRs performance in the overlay. Essentially, the near real-time performance of all XOMRs is available for use in performance optimization of the overlay.

This section defined the throughput and loss ratio in terms of *n*-degree where degree means the number of replicated messages that an XOMR might be required to make given the topology of the overlay. Using the measurement capability of the XOMR, it is

then possible to use the information to define performance in terms of *n*-degree. The local XOMR then can make decisions about its role in the overall overlay. For example, the XOMR can implement a decision process that says it is able or unable to support a relay function between XOMRs or accept a new relay function given the current *n*-degree of the node. The XOMR also could use the information to accept or deny new traffic flows from the local subnet or offer a reduce service with less performance guarantees. The idea is to maximize available capacity in the overlay. This is consistent with the proposed approach for the XOMR, where each XOMR is essentially a proxy agent that has the ability to replicate messages and forward to the appropriate local host or other XOMR node in the overlay.

There are at least two possible alternatives to use this information for construction and management for the overlay. The first is to implement a two-step construction process similar to the Scattercast protocol described in Section 3 and the second is to add a parameter to the current prototype implementation of Dijkstra's algorithm that reflects the ability or willingness of an XOMR to support additional capacity in support of the overlay network.

Recall that Scattercast uses a two step process to build the overlay. It first builds a mesh of nodes randomly and is able periodically to locate the best nearest neighbor if there is one better than the current. Scattercast then runs a distance vector algorithm to form a minimum routing tree.

The second alternative is to take advantage of the current prototype use of Dijkstra's algorithm directly by adding a constraint for degree. The current implementation builds a

distribution tree given known other XOMRs in the overlay. Information about other nodes is obtained by XOMRs exchanging information. Implementation of a registry is part of the proposed architecture and will provide services to assist this process in the future. An attractive concept would be to modify Dijkstra's algorithm to include an index for the XOMR's degree or capacity to increase degree. Kwon [Kwon02] provides an outline of an algorithm that could be modified to perform this function. Boehm [Boeh02] developed a variation of Dijkstra's algorithm for constraint routing, using heuristics to reduce computational complexity.

The second key observation is that the arrival process is representative of ON/OFF and never zero with messages sizes deterministic in nature. The previous section demonstrated that this can be represented analytically as an M/D/1 queue model. The implication is that we expect better performance of the queueing mechanism than if the messages size where exponentially distributed. In fact, the M/D/1 queuing system has $\frac{\rho^2}{2*(1-\rho)}$ fewer messages in the system than the M/M/1 queuing system [Klei75]. This means that we should be able to operate the XOMR with higher queue threshold values for the same throughput than might be available if the XOMR where an open network router or lower layer service router. However, if operating at a higher threshold, then, a mechanism to handle bursts of messages may be required.

There are two possible approaches to handle higher arrival rates or bursts. Both approaches are included in the proposed architecture. The first is to use a protocol like the Selectively Reliable Protocol (SRMP) as an interface to the XOMR and the second is to

implement a priority queueing mechanism and offer two classes of service; Class B with no priority and Class A with priority.

While the SRMP is designed to offer a selectively reliable service, it in effect serves to function as a flow control mechanism by giving a higher grade of service to messages that are marked for higher reliability. For the case of two class queuing, the service gives priority to the higher class when congestion occurs. The lower class messages are dropped if the queue overflows.

IV Summary

The design of the architecture is focused on meeting the many-to-many multicast service requirement of distributed simulations with high confidence in the service performance. This Section has related key aspects from the observation of live simulations and key performance characteristics of the current prototype to architecture design considerations. Included was development of a link between the proposed analytical model and evaluation of an early prototype of the XOM overlay. The purpose is to validate the analytical model and serve to form a basis of key features in the proposed architecture.

Part II of this section presented the results of laboratory tests of the current XOMR prototype conducted in the GMU C3I Laboratory and related key measurements to the concept of a node degree where degree means the number of message replications are to be performed for an arriving message. Part III discussed key aspects of the proposed

architecture that were influenced by results from laboratory tests as well as from the study of live simulations.

SECTION 7 CONCLUSIONS, CONTRIBUTIONS, AND RECOMMENDATIONS FOR FUTURE RESEARCH

I Introduction

Many-to-many multicast transmission is an essential network capability for distributed real-time virtual simulation. However, many open issues make network-layer multicast impractical over the Internet leading to consideration of alternative strategies to support the distributed real-time simulation application environment. Previous sections developed a description of the problem, proposed an alternative strategy and provided evidence to support the conclusion that it is feasible to provide multicast services for the environment based on the concept of a network overlay. This section discusses the conclusions of the research and summarizes the unique contributions that resulted from the work. The section concludes with recommendations for future research that can serve to extend this effort.

II Conclusions

This research has demonstrated that it is feasible for host-based or end system overlay multicast to provide multicast services to real-time distributed simulation across an open network. The results further indicate that implementing an overlay multicast strategy allows the distributed simulation environment to maintain some independence of the

underlying network services thus allowing use of lower layer Internet protocols. To demonstrate the feasibility, the following items were accomplished:

- Detailed studies were conducted of three live simulations. The simulation environments where analyzed by using OPNET Modeler where actual traces of message behavior was captured and used to characterize the requirements for overlay multicast performance. These studies provided the baseline for comparison of expected performance for the overlay approach.
- A message traffic load generator was implemented based on the results of the simulation studies and used as the control load for laboratory studies of overlay system performance.
- An analytical model was developed to aid in the understanding of system behavior and providing a statistical basis for predicting performance. The model was validated by comparison to message load studies in the laboratory using OPNET Modeler analysis tools for data capture.
- Using the message traffic generator and an overlay multicast protocol prototype, detailed performance studies where conducted in a laboratory with various network configurations across open network Internet protocols. These studies were used to validate the analytical model and form the basis for predicting expected performance of an operational overlay multicast protocol.

III Unique Contributions of this Research

The conceptual development and demonstration of the fundamentals for a host-based multicast protocol represents an important contribution to knowledge of how best to support many-to-many communications for real-time distributed virtual simulations in an open network. The research demonstrated the power of the overlay approach for providing networking flexibility for real-time distributed simulations by enabling users of these applications to network without the need for costly private networking. This approach helps to expand the real-time distributed simulation user base since the concept could extend to individual personal computers (PCs) acting as relay nodes, enabling low-cost end-to-end multicast schema.

The research addressed the issue of scaling the performance of the overlay. A methodology was defined and demonstrated for studying the effects of scaling based on the idea of node degree where degree means the number of replicated messages or channels to neighboring overlay nodes. The application of this idea forms the basis for defining a capacity of an overlay node in simple terms that can be used by a routing algorithm in the dynamic formation of an overlay network.

The research made a unique contribution in the characterization of message load generated by real-time virtual distributed simulations. Three live simulations were studied which resulted in detail characterization at the simulation federate level as well as the aggregated flow of message traffic in a very large simulation with 110, 000 federates operating across a private network built over the DREN.

The results of these studies enabled the formulation of an analytical model to describe the environment. This was a significant contribution as this model provides a statistically relevant model that can be directly used for managing and deploying many-to-many overlay multicast network. The analytical model provides the ability to forecast expected performance based on real-time measurement of aggregated message flow through an overlay node.

The primary outcome of this research was to develop a high level architecture of a proposed overlay multicast protocol that would provide many-to-many multicast services for real-time distributed virtual simulation. This architecture was developed based on results of the traffic studies of live simulations and the subsequent development of analytical model as well as prototype testing in the laboratory. The proposed architecture recognizes that underlying networks may have a wide range of network capacities and capabilities yet provides a multicast service to higher layer applications that require this capability across open networks. The proposed approach includes consideration for reliability by providing two classes of services such that the application may specify message handling a under congested network conditions.

IV Recommendations for Future Research

The original purpose of this research was to validate and demonstrate the feasibility of using overlay multicast to support a very demanding communications requirement. The outcomes of the current research have provided confidence that the approach is valid. As the work moves to the next level in experiments and development, it is clear that there

are many challenges remaining. Research focus should continue to be on efficiency and scalability as well as overlay management mechanisms that respond to the dynamic nature of underlying open networks as well as the dynamic nature of the application environment. The next steps for consideration in continuing the work are:

- Validate the approach by using the prototype in a live wide area network simulation requiring performance levels that stress the capabilities of the XOMR.
- Establish a testing environment where evaluation of specific routing algorithms can be performed. The results of this testing will refine the routing features of the overlay and provide for improved optimization of overlay network performance.
- Research and add security features to the prototype for protection from denial of service attacks and related intrusions and implementation of capabilities to support end-to-end information protection features.
- The current prototype employs services over UDP. This should be expanded to include a concept of TCP tunnels for improved security and for those applications that have streaming data that desire more reliability than UDP. The study of TCP should include how to manage the slow start feature of TCP so as not to hinder performance.
- Continue refining the protocol by prototyping repeatedly at increasing levels of sophistication and optimization.
- Refine and continue integration with early prototypes of web-services used in distributed real-time simulations.

- Research and prototype registry services for the overlay as part of an over all improvement in overlay management capabilities.
- Research and prototype the ability to provide for two priority classes, “priority” and “best effort,” in cases where aggregate application throughput requirements are at or above the capacity provided by the underlying network.
- Research how an overlay multicast protocol could take advantage of commercial efforts to improve underlying network QoS and use services such as Multiprotocol Label Switching (MPLS).

APPENDIX B END SYSTEM MULTICAST DEFINITION

This Appendix describes how End System Multicast works and relates it to classic IP multicast.

Distance Vector Multicast Routing Protocol (DVMRP) is the oldest multicast protocol and was first defined in RFC 1075 [Mill99]. Using DVMRP, data is delivered from the source to the receivers using an IP multicast tree of the shortest paths from each receiver to the source. Figure B-1 presents an IP multicast tree constructed using DVMRP. Routers A and B receive a single copy of the packet and forwarded it along multiple interfaces. However, only one copy of the packet is sent over any physical link. In addition, packet delay is the same to all receivers as though packets were sent directly by unicast.

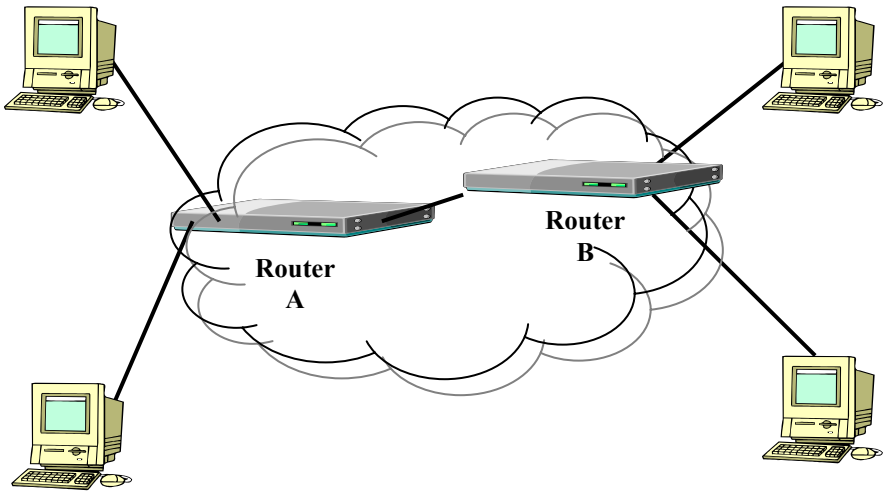


Figure B-1: IP Multicast Tree resulting from DVRMP

We can expand the representation of the multicast function in this simple example [Figure B-2] to a complete graph where every pair of nodes is the set of ends of an edge representing the virtual flow of packets [Bert98]. DVMRP then becomes nothing more than solving the minimum spanning tree problem for this complete graph.

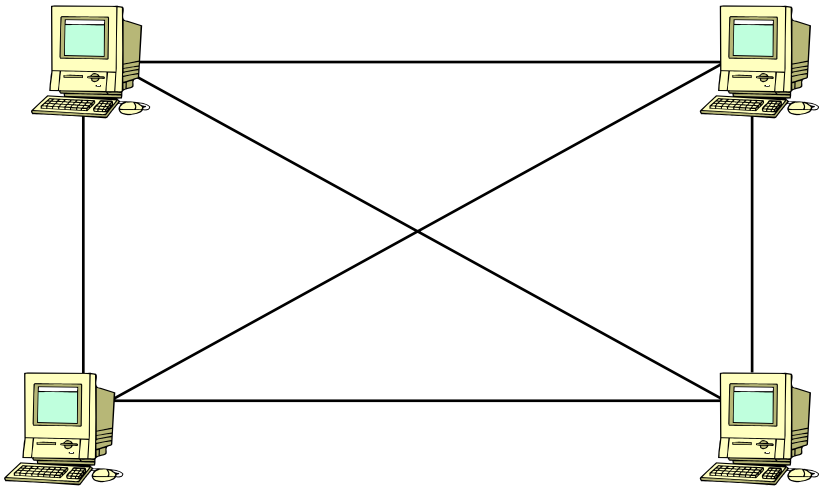


Figure B-2: Complete Graph connecting all Nodes

If we now remove the multicast function from the routers A and B in the network and build a new tree based on the end systems, then we get a spanning tree as presented in Figure B-3. Unlike the DVMRP example, the underlying physical path of the packets is now represented as in Figure B-4. The result is that we are now re-using some of the physical paths twice and we have added a slight amount of delay.

Figure B-3: Spanning Tree of all Nodes

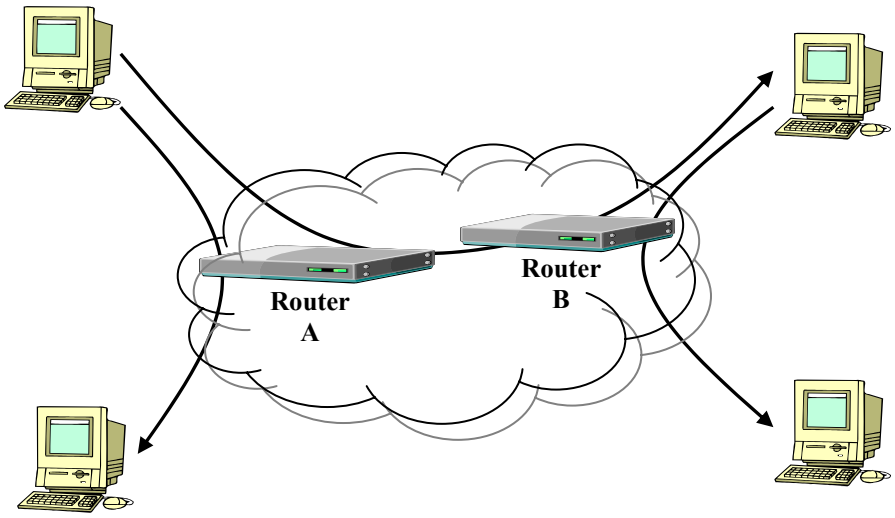


Figure B-4: Physical Path of Packets across Spanning Tree

The End System multicast concept can be generalized to include nodes at the edge of a network where proxies can use the LAN multicast functionality [ChuY00]. These proxies can also act as a router and forward packets on behalf of other nodes [Figure B-5].

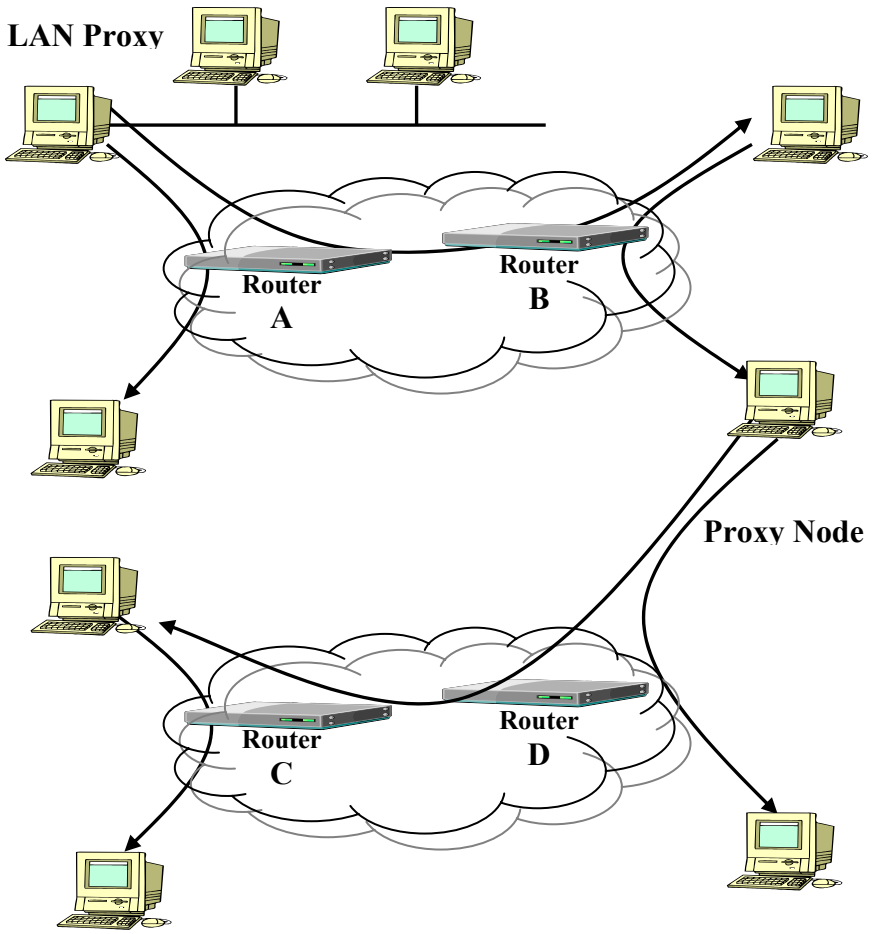


Figure B-5: Proxy Nodes in an Overlay

APPENDIX C XOMR PROTOTYPE

XOMR Prototype Description

This document provides a brief description of the current prototype of the Extensible Modeling and Simulation Framework [Brut02] Overlay Multicast Relay (XOMR). The XOM is an overlay multicast protocol designed to support many-to-many multicast for real-time distributed visual simulations. The objective is to provide multicast service over a unicast network environment using UDP. From the multicast sender and receiver's point of view, each XOMR looks like an IP layer multicast router.

The XOMR performs as a multicast "relay agent" for any application located on the same subnet as an instance of the XOMR as indicated in Figure C-1. For each subnet that participates in the multicast group communication, there must be a host running a XOMR on the subnet. The XOMR listens to the local LAN, for each multicast packet generated within its local subnet and it will forward this multicast packet to the downstream XOMR(s) according to its multicast tree, by using unicast. Figure C-2 presents the concept and indicates the concept of group aggregation efficiency gains by using the overlay multicast. The partner XOMR will then multicast the packet to the destination local LAN, and keep on forwarding the packet to other XOMRs if necessary.

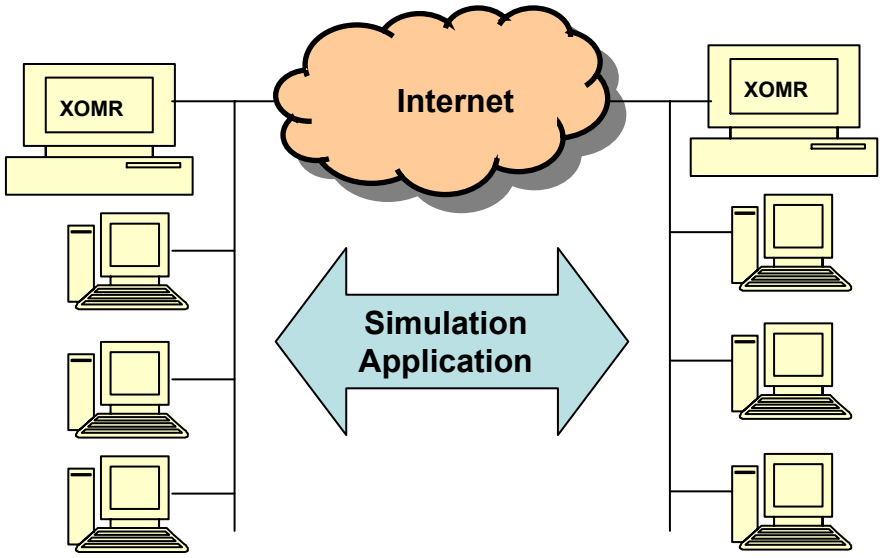


Figure C-1: XOMR Service on a Subnet

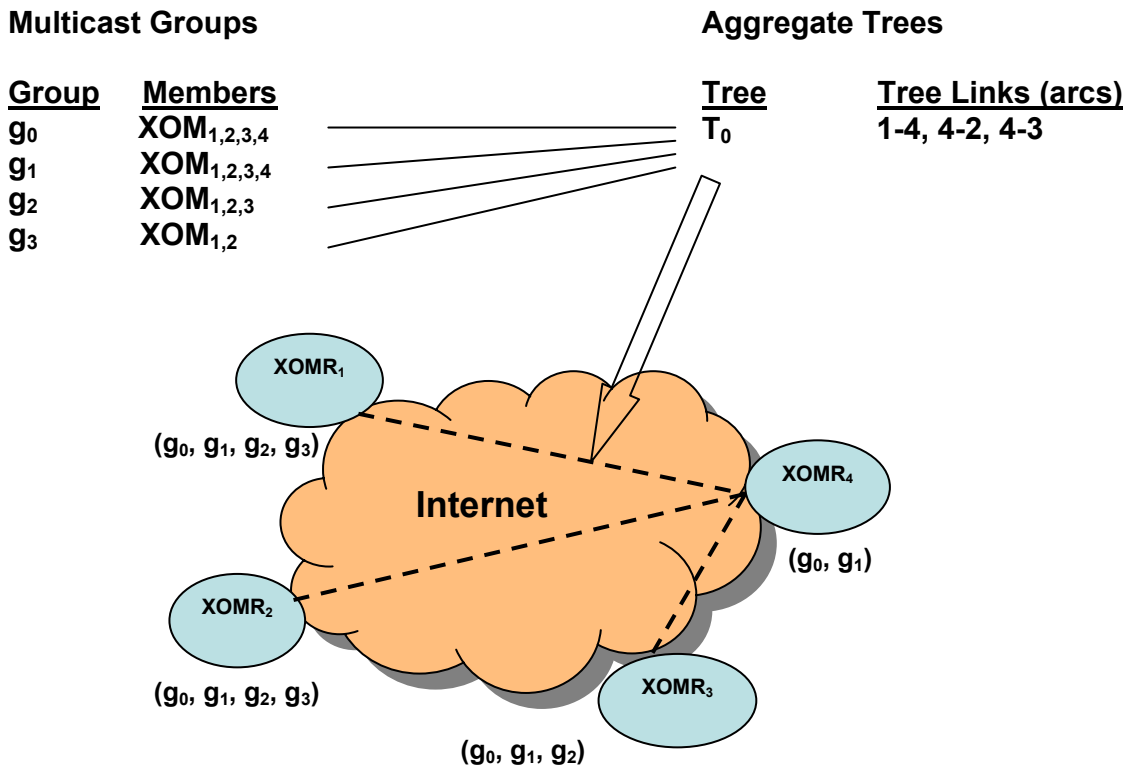


Figure C-2: Group Aggregation Ovelay

Key elements of the current XOMR prototype:

- Group Management

  The group addresses and UDP port used by the XOMR are specified as command line arguments. IGMP is not implemented in this version of code

- Partner Discovery

  The partner XOMR's IP addresses is specified as command line arguments.

- Network Measurement

XOM uses delay as the routing metric. A virtual mesh composed of all the XOMRs is established. This mesh is a concept of the transport layer. The current version establishes a full mesh, and every XOMR maintains this mesh topology in a matrix that also includes the measured network delay between XOMRs.

Each XOMR periodically sends Ping message to all partner XOMRs it knows. Each XOMR will echo Ping messages received from other XOMRs and use the response to measure round trip times RTT) as the measure for delay. Periodically, each XOMR will flood its measured RTT values to all its XOMR partners. At last, every XOMR will keep a delay matrix of RTT value between every two XOMs.

- Multicast Forwarding Tree calculation

  Periodically, each XOMR will calculate a source specific multicast tree according to its delay matrix. The tree is rooted at the source XOMR. The prototype uses Shortest Path First (SPF) tree algorithm.

- Routing Update

  Periodically, each XOMR will flood its multicast tree to all other partner XOMRs. Each XOMR will keep *n* multicast trees, where *n* equals to the total number of XOMRs.

- Packet replicating and Forwarding

  When a XOMR receives a multicast data packet from its local LAN, it will encapsulate it, replicate it, and forward it to the downstream neighbors according to its own multicast tree. When a XOMR receives a UDP data packet from

another XOMR, it will check to see from which source XOMR the UDP packet was originally generated (but not the upstream neighbor). It then checks the locally stored multicast trees for that source XOMR to replicate the packet and forward it to other downstream XOMRs if there is any. Meanwhile, the XOMR will de-capsulate the UDP packet and multicast it to its local LAN.

Prototype Software Design

The XOMR was first implemented in JAVA. Then, to improve the performance, the Packet replicating and forwarding modules were implemented in C++. The control module remains implemented in Java. The JAVA and C++ modules were integrated by using Java JNI callback.

Figure C-3. presents the design structure of the JAVA version XOMR implementation. The JAVA version is composed of the following four major classes:

- *public class XOM*; This is the main class and creates all the threads and initializes the data structures. This class also provides general methods and auxiliary functions, such as network layer address format conversion, encoding and decoding, searching, etc.
- *public class XOMMulticastRouter extends Object implements Runnable;*This is the forwarding engine. This class replicates and forwards the packets. It listens to both the local LAN and other XOMRs. It stores the forwarding table

in its member variable *XOMMulticastRouter*.*forwardingTables:* The forwarding table is periodically updated by class *XOMRoutingTable*.

- *public class XOMRoutingTable implements Runnable;*This class performs the real routing work. It processes the delay information and multicast tree information received from other XOMRs. It then calculates its own multicast tree according to this information, and updates the forwarding table in class *XOMMulticastRouter*:
- *public class XOMStatistics implements Runnable;* This class measures the RTT values to all the other XOMRs. It also does the statistics work, such as total number of packets sent and received.

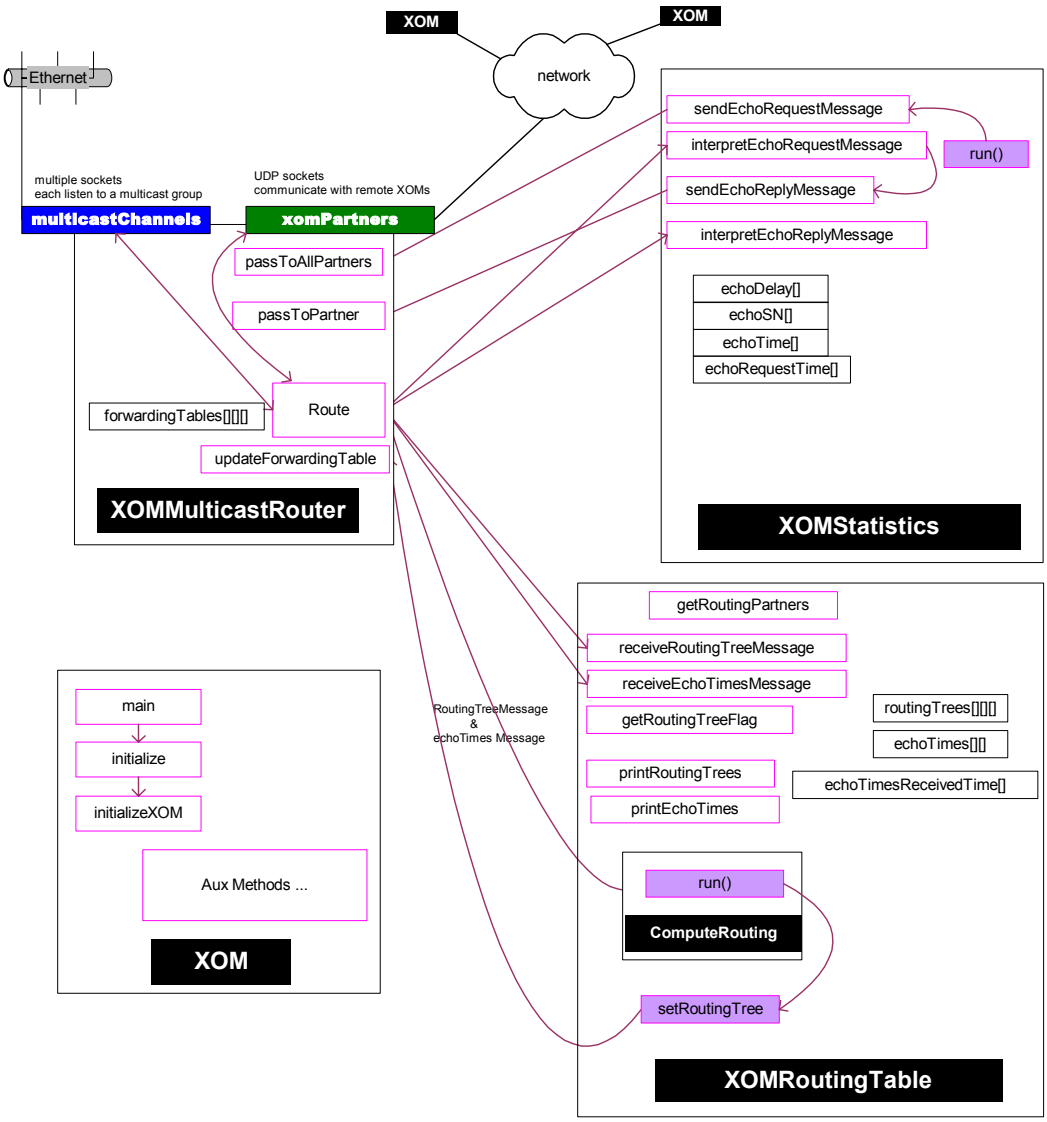


Figure C-3: JAVA Version of XOMR

The C++ version code (Figure C-4.) implements the *XOMMulticastRouter* class in C++. The other three classes are still in JAVA. The C++ module communicates with

the JAVA module by using JNI (Java Native Interface). All the data packets are processed within the C++ module, which includes sending/receiving, encapsulation/de-capsulation, and forwarding. All the control messages are passed to the JAVA module. The C++ module keeps a forwarding table, which is updated by the JAVA module. The following diagram shows the C++ version design structure:

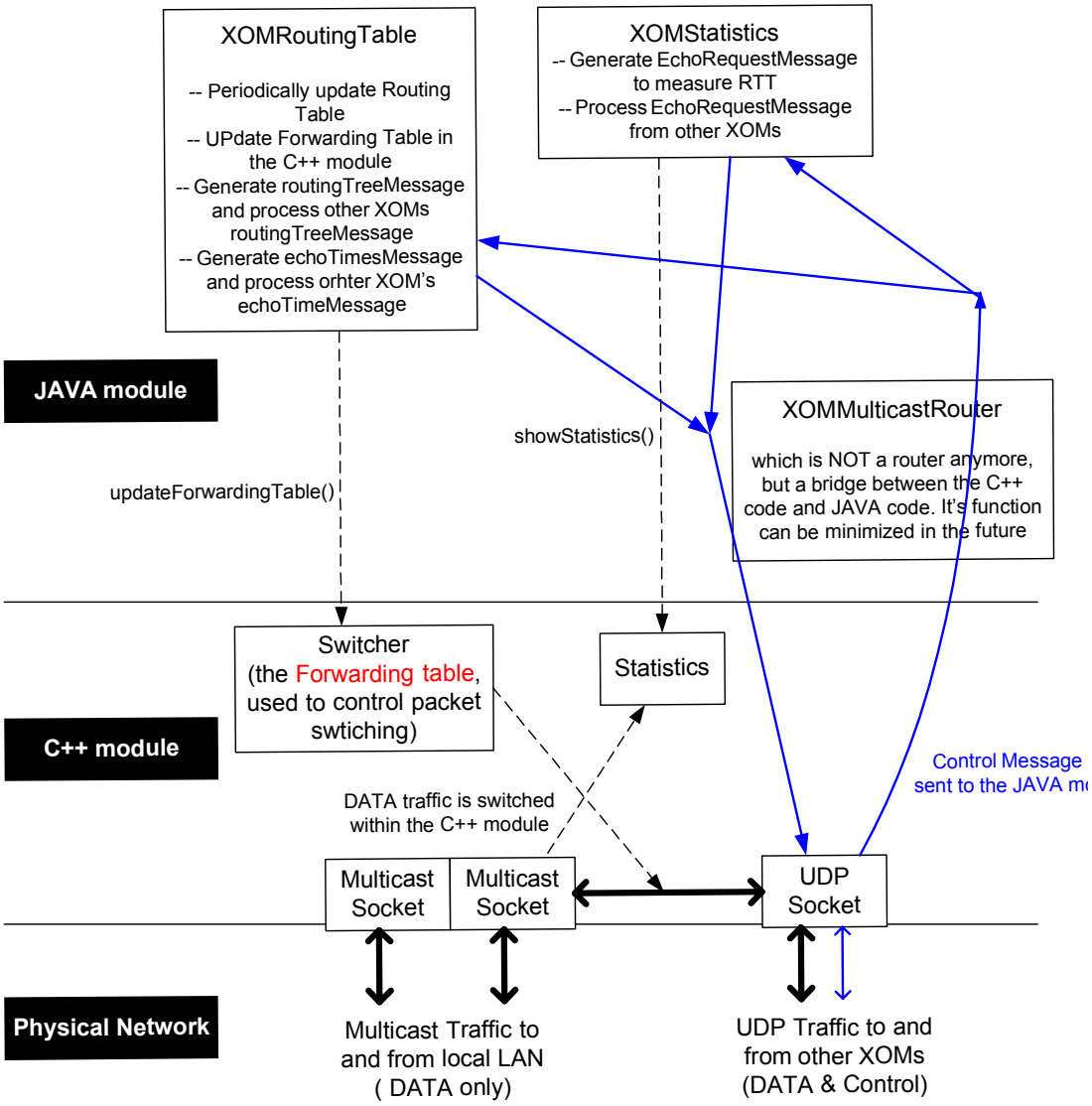


Figure C-4: C++ Version of XOMR

Client Interface Specification

The client refers to the multicast sender and receiver. The XOMR provides a transparent service to the client. The client can simply treat the XOMR as an IP layer multicast router. Currently, the XOMR does not speak IGMP, however, IGMP is planned for future versions.

The UDP port used by XOM for communications other is specified by XOM.XOM_PORT. Currently, it is default set to UDP port 4785.

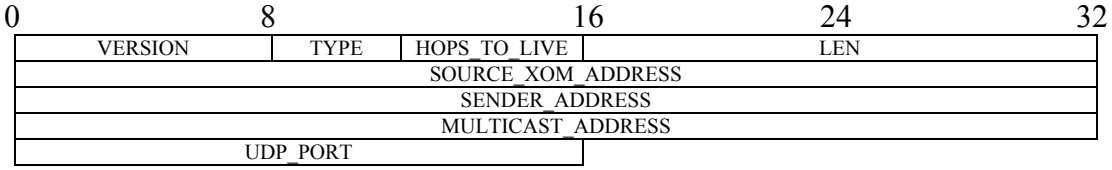
- Command Line Arguments

java XOM <registryAddress> <numberOfMulticastGroups> <numberOfPortsPerGroup> <lowestMCAddress> <lowestPort> <routingUpdateInterval> <thisSubnetMaskBits> <debugIndexThisXOM> <useTCP> [partnerXOMAddress1, partnerXOMAddress2, ...]

registryAddress	- InetAddress of registry, 0 if none
numberOfMulticastGroups	- count of groups/ports we will support
numberOfPortsPerGroup	- count of ports each group will support (non-verlapping)
lowestMCAddress	- first group address to multicast from the subnet, dotted decimal notation (other addresses follow in sequence)
lowestPort	- first UDP port to multicast (each address will get one port in sequence)
routingUpdateInterval	- time in ms between routing updates (default 10 s)
thisSubnetMaskBits	- number of bits used for routing in subnet address (default 24)
debugIndexThisXOM	- 0 for operation; else index of this XOM in development
useTCP	- 0 for UDP tunnels, 1 for TCP tunnels
partnerXOMAddresss	- zero to MAX_PARTNERS addresses, in dotted decimal format, to be used as partners without checking the registry

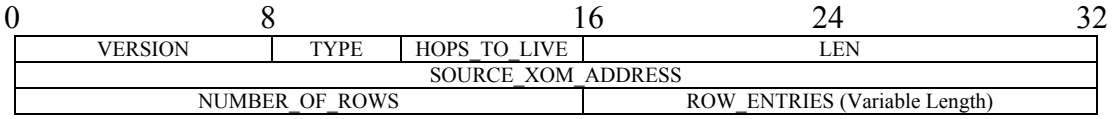
Packet Format

- DATA:



VERSION: 8 bits, 0x 2
TYPE: 4 bits, 0x 1
HOPS_TO_LIVE: 4 bits, the number of hops can be relayed by XOMs
LEN: 16bits, the packet length including the header
SOURCE_XOM_ADDRESSS: 32bits, the IPv4 address of the XOM that generate this UDP packet
SENDER_ADDRESS: 32bits, the IPv4 address of the end host that generate the multicast packet
MULTICAST_ADDRESS: 32bits, the multicast address used by the end host
UDP_PORT: 16 bits, the UDP port used by the end host

- ROUTING_TREE:

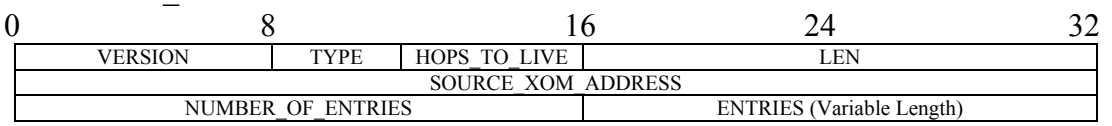


For Each ROW_ENTRY:

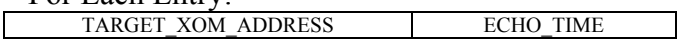
For Each COLUMN_ENTRY:

VERSION: 8 bits, 0x 2
TYPE: 4 bits, 0x 2
HOPS_TO_LIVE: 4 bits, the number of hops can be relayed by XOMs
LEN: 16bits, the packet length including the header
SOURCE_XOM_ADDRESSSS: 32bits, the IPv4 address of the XOM that generate this routing information
NUMBER_OF_ROWS: 16bits, the number of rows of the routing_tree matrix
ROW_ENTRIES: variable lengths, may be composed of multiple ROW_ENTRYs
PARTNER_XOM_ADDRESS: 32 bits, the IPv4 address of the XOM corresponding to the row
NUMBER_OF_COLLUMNS: 16 bits, number of downstream partner XOMs
COLUMN_ENTRIES: variable length, may be composed of multiple TARGET_XOM_ADDRESS
TARGET_XOM_ADDRESS: 32 bits, the IPv4 address of the downstream XOM

- ECHO_TIMES:



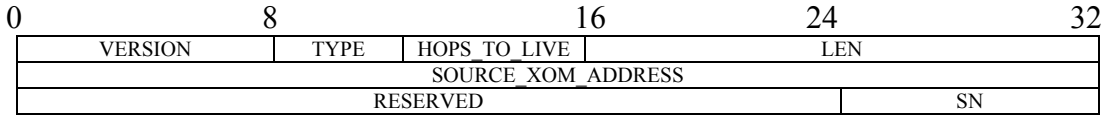
For Each Entry:



VERSION: 8 bits, 0x 2
TYPE: 4 bits, 0x 3
HOPS_TO_LIVE: 4 bits, the number of hops can be relayed by XOMs

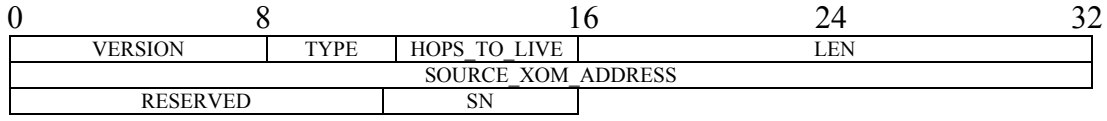
LEN: 16bits, the packet length including the header
SOURCE_XOM_ADDRESSS: 32bits, the IPv4 address of the XOM that generate this echo time list
NUMBER_OF_ENTRIES: 16 bits, the number of entries in the echo time list
ENTRIES: variable length, may be composed of multiple entries
TARGET_XOM_ADDRESS: the IPv4 address of the target XOM
ECHO_TIME: the RTT value between the SOURCE_XOM_ADDRESS and TARGET_XOM_ADDRESS

- ECHO_REQUEST:



VERSION: 8 bits, 0x 2
TYPE: 4 bits, 0x 4
HOPS_TO_LIVE: 4 bits, the number of hops can be relayed by XOMs
LEN: 16bits, the packet length including the header, always equals to 0x12
SOURCE_XOM_ADDRESSS: 32bits, the IPv4 address of the XOM that generate this echo request
RESERVED: 24 bits
SN: 8 bits

- ECHO_REPLY:



VERSION: 8 bits, 0x 2
TYPE: 4 bits, 0x 5
HOPS_TO_LIVE: 4 bits, the number of hops can be relayed by XOMs
LEN: 16bits, the packet length including the header, always equals to 0x12
SOURCE_XOM_ADDRESSS: 32bits, the IPv4 address of the XOM that generate this echo reply
RESERVED: 24 bits
SN: 8 bits

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