An Improved Active Network Concept and Architecture for Distributed and Dynamic Streaming Multimedia Environments with Heterogeneous Bandwidths

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to my family

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List of Abbreviations

ADN	Anycast Domain Name
AGLP	Adaptive Gateway Location Protocol
ALAN	Application Level Active Network
AN	Active Network
ARM	Active Reliable Multicast
ATM	Asynchronous Transfer Mode
BER	Bit Error Rate
bps	bit per second
BW	Bandwidth
Codec	Coder Decoder
CDN	Content Distribution Network
ConCEPT	Content Cache with External Proxylet Transcoding
CSCW	Computer Supported Collaborative Work
DAAD	Deutscher Akademischer Austausch Dienst
DiffServ	Differentiated Services
DNS	Domain Name server
DPS	Dynamic Protocols/Proxylets/Proxy Servers
EEP	Executive Environment for Proxylet
FANS	Friendly Active Network System
fps	Frame Per Second
GoP	Group of Pictures
HTML	Hypertext Markup Language
HTTP	Hyper Text Transfer Protocol
IETF	Internet Engineering Task Force
IntServ	Integrated Services
IP	Internet Protocol
ITS	Institute for Telecommunication Science
JMF	Java Media Framework
JPEG	Joint Photographics Experts Group
JVM	Java Virtual Machine
kbps	kilobits per second
LAN	Local Area Network
Mbps	Megabits Per Second
MeTL	Media Tracking List
MmTL	Member Tracking List
NTIA	National Telecommunication and Information Administration
PDA	Personal Digital Assistants

PLR	Packet Loss Ratio
QoS	Quality of Service
RFC	Request For Comment
RLM	Receiver-driven Layered Multicast
RMI	Remote Method Invocation
RON	Resilient Overlay Network
RSVP	Resource Reservation Protocol
RTP	Realtime Transport Protocol
RTCP	Real Time Control Protocol
RTSP	Real Time Streaming Protocol
SIF	Standard Intermediate Format
SLP	Service Location Protocol
SRP	Service Registration Point
UDP	User Datagram Protocol
url	Universal Resource Locator
WWW	World Wide Web
VDP	Video Datagram Protocol

Chapter One

Requirements for Streaming Multimedia over Environments with Heterogeneous Bandwidth Capacities

1.1 Introduction

A problem in today's Internet infrastructure may occur when a streaming multimedia application is to take place. The information content of video and audio signals that contain moving or changing scenes may simply be too great for Internet clients with low bandwidth capacity if no adaptation is performed.

To fairly reach clients with various bandwidth capacities some works such as receiver-driven multicast [CAN96] and resilient overlay networks (RON) [AND01b, AND01b] have been developed. However these efforts mainly call for modification on router level management or place additional layer to the current Internet structure, which is subject to the lengthy standardization process. An improved active network approach for distributed and dynamic streaming multimedia environment with heterogeneous bandwidth, such as the case in the Internet has been developed. The concept and architecture requires no changes in router level management and put no additional requirement to the current Internet architecture and, hence, instantly applicable.

Friendly active network system (FANS) [RAM00, RAM01, RAM02, RAM03a, RAM03b, RAM03c] is a sample of this approach. Adopting an application level active network (ALAN) mechanism, FANS participants and available media are referred through its universal resource locator (url). The system intercepts traffic flowing from source to destination and to perform media post-processing at an intermediate peer. The process is performed at the application level instead of at the router level, which was the original approach of active networks.

In comparison with ALAN, FANS possesses two significant differences. From the system overview, ALAN requires three minimum elements: clients, servers, and dynamic proxy servers. FANS, on the other hand, unifies the functionalities of those three elements. Each of peers in FANS is a client, an intermediate peer, and a media server as well. Secondly, FANS member's tracking system dynamically detects the existence of a newly joined computers or mobile device, given its url is available and announced. In ALAN, the servers and the middle nodes are priori known and, hence, static. The application level approach and better performance characteristics distinguished also our work with another similar work in this field, which uses router level approach.

This chapter discusses the problem statement of this dissertation. The ultimate problem is the difficulties in providing sufficient QoS to deliver multimedia services. The current Internet conditions can cause the degradation of QoS in delivering multimedia applications and affecting the three QoS parameters namely bandwidth, packet loss and delay. Among the sources of QoS degradation are the heterogeneity of Internet bandwidth at the client sites and the diversity of the intensity traffic throughout the global Internet.

1.2 Multimedia Demands and Internet Traffic Density and Intensity

The Internet and its applications have emerged since the four universities, i.e. University of California Los Angeles (UCLA), Stanford Research Institute, University of California Santa Barbara (UCSB) and the University of Utah, each hooked up a Honeywell DDP-516 mini computer with 12 K memory in 1969 [ZAK03]. Internet does not provide mechanisms to guarantee an upper bound on end-to-end delay or a lower bound on available bandwidth. Consequently, the quality of delivered services is neither controllable nor predictable.

No one would have then thought that quality of service (QoS)-intensive multimedia applications such as streaming of video, audio and videoconferencing over the Internet becoming reality. However, lack of support for QoS has not prevented rapid growth of real-time streaming multimedia applications and this is expected to continue and an expert in International Telecommunication Union (ITU) [KEL01] stated that by 2005 the Internet might be available throughout the world and it is used primarily for multimedia streaming applications.

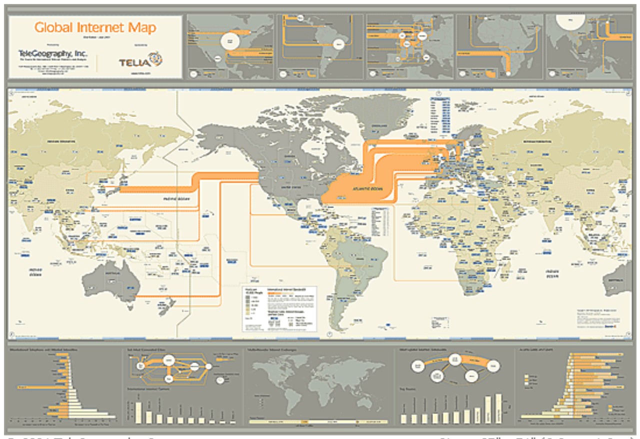
An obvious parameter influencing the QoS of Internet applications is bandwidth. In terms of multimedia applications, if there is not enough bandwidth available, then there is no way the data can get from source to destination without the video and audio streams having to rebuffer. In addition, since it is not guaranteed that each of the Internet packets reaches their destination then the other factor is packet loss. Packet loss occurs if the incoming data link speed is faster than the capability of router or end station, whose limited buffer, to forward the packet to the next router or to pass the packet to the higher level. Buffering, furthermore, creates additional delay to the existing propagation delay and router delay. Too long delay, in turn, makes full duplex conversation difficult to understand. Hence, another parameter for Internet QoS is the delay.

Between year 2000 and 2001 the averaged international Internet bandwidth grew 278 percent. The Internet's global topology, as shown in Table 1.1, is growing in uneven bursts. For example, Latin America's international connectivity grew by almost 480 percent to 16.1 Gbps [TEL02].

Table 1.1 International Internet Bandwidth by Region, 2000 – 2001 [TEL01]

Region	2000 Mbps	2001 Mbps	% Growth
Africa	649.2	1,230.8	89.6%
Asia	22,965.1	52,661.9	129.3%
Europe	232,316.7	675,637.3	109.8%
Latin Am.	2,785.2	16,132.5	479.2%
U.S. & Can.	112,222.0	274,184.9	144.3%
© TeleGeography, Inc. 2001 Note: Data as of mid-year.			

It is clear that high-speed bandwidth accesses are not available for everybody. With optical fibres and Dense Wavelength Division Multiplexing (DWDM) technologies a very high bandwidth, up to terabits per second, is achievable in the core network. Internet backbone providers can purchase capacity on these systems in larger quantities and at lower prices than previously available. However, the end users still have diversity accesses ranging from 56 Kbps to 2 Mbps and these last miles problems will be the case for many years. For example, while in the developed countries such as USA, Germany, and UK the capacity of Internet bandwidth to and from Universities and institutions is around 650 Mbps or more, in some other developing countries is still around 640 Kbps or even worse.



© 2001 TeleGeography, Inc. Size = 37" x 51" (0.9 m x 1.2 m)

Figure 1.1 World Internet Map (2001) [TEL01]

The capacity of Internet links between Latin America and the rest of the world has a faster rise than for any other region, and has quintupled over the past year. The United States still plays a central role in Internet infrastructure as the world's main hub. As in mid-2001, over 80 percent of international Internet capacity in Asia, Africa, and South America still connected directly to a U.S. However, most countries have become less dependent on the U.S. as a switching station. This observable fact is depicted in Figure 1.1.

Figure 1.2 depicts calculation of international Internet bandwidth per person in developing countries. People in Chile have the highest available bandwidth capacity whereas India surprisingly comes after most of the countries in this region.

Due to the huge differences in bandwidth capacity at the access points (e.g., to countries and to continents), the intensity and the density of the Internet traffic at the different regions in the world are significantly diverse as well. The Internet traffic flows between USA and European countries is the most intense one but not necessarily the most congested one due to its broadest capacity.

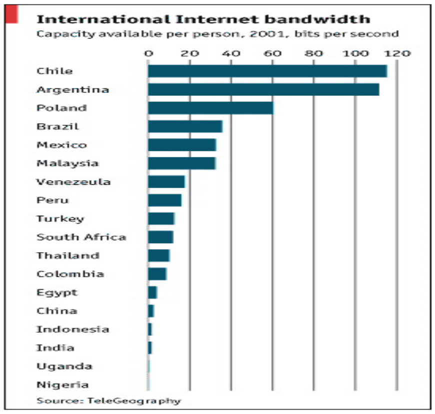


Figure 1.2 The Internet Bandwidth per Person in the Developing Countries [TEL01]

Countries in USA and European have the potential to become streaming multimedia region. However, only limited countries in Asia such as Japan, Australia, South Korea and Singapore would be able to support the streaming applications. Other countries could be considered to have less bandwidth for streaming applications. The diversity of bandwidth capacities in Asian region is shown in Figure 1.3.

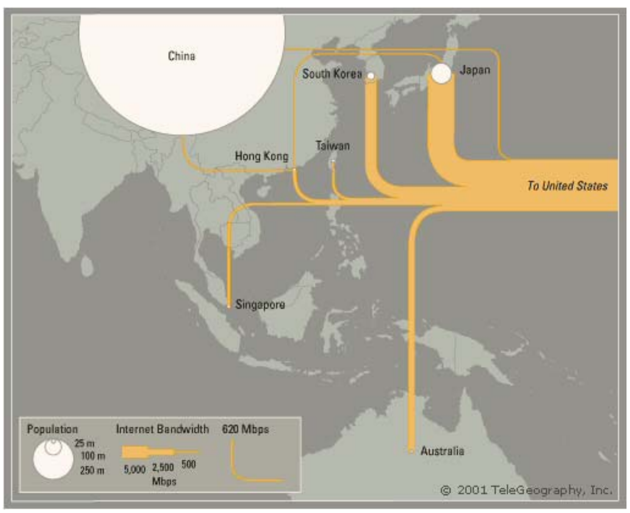


Figure 1.3 The Bandwidth Heterogeneity in Asia Pacific [TEL01]

The intensity of Internet traffic in Africa is the lightest one. With 13% of the world's population Africa takes only 0.15% of international Internet connections but might already be the most congested one due to its bandwidth capacity. This phenomenon is shown in Figure 1.4.

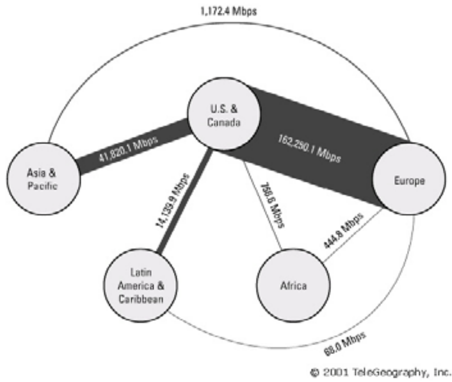


Figure 1.4 Interregional Internet Traffic [TEL01]

As a result of the impressive construction, connections between European cities are now better than inner-American Internet hook-ups. Many European connections are broadband, but not yet being used at full capacity. As the result, the connection from and to European cities is not as intense and congested as in USA.

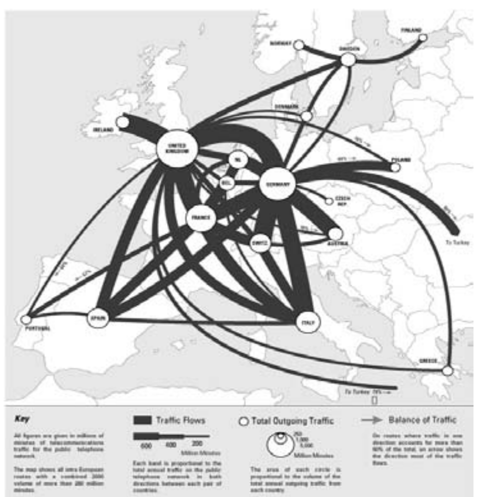


Figure 1.5 Internet Traffic Flow in Europe [TEL01]

The largest inter-continental connections exist between the United States and Europe with a data transfer capacity of around 162 gigabytes per second. The connections between the United States and Asia, at 42 gigabytes per second, are at a distant second. The United States-South America connection runs at 14 gigabytes per second, while Africa is connected to the rest of the world at less than one gigabyte per second. The Internet traffic intensity for European, Latin American, and Asia regions are shown in Figure 1.5, Figure 1.6 respectively.

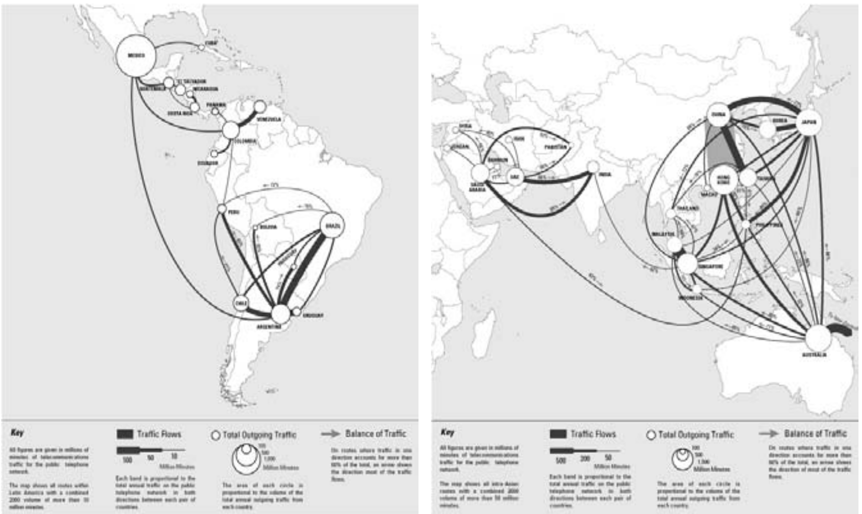


Figure 1.6 Internet Traffic Flow in Latin America (Left) and Asia (Right) [TEL02]

Traffic redirection in the Internet is existent. Internet traffic takes multi-hops path from origin to destination. The Internet traffic to countries with lower bandwidth capacity is delivered through non-congested regions or regions with very high bandwidth capacity like USA and Europe. Most of the traffic is delivered through USA as main switching point. Observation of this dissertation work (APPENDIX A to APPENDIX F) shows that only traffic directed from Germany to Japan, Australia and Singapore that do not take USA as one of its hop. Other destination such as Indonesia, Malaysia and other countries in Africa and Latin America, based on observation of this dissertation and the report from Telegeography Inc. [TEL02], are connected directly to cities in USA.

The congestion that frequently occurs during transmission is mainly due to the bottleneck regions, which generally takes place at the access points to the countries with poor bandwidth. The problem, hence, is not at the backbone network, which currently can take up to several Terrabits per second, but at the access points to clients.

As the above information and statistics shown, it is obvious that to be able to deliver streaming multimedia applications to continents and countries with poor bandwidth capacities certain mechanisms are required. The mechanisms should improve the flexibility of the stringent streaming multimedia requirements. These may include the quality of presentation picture, the size of presentation frame and the number of frames per second.

1.3 Internet and Environments with Heterogeneous Bandwidth Capacities

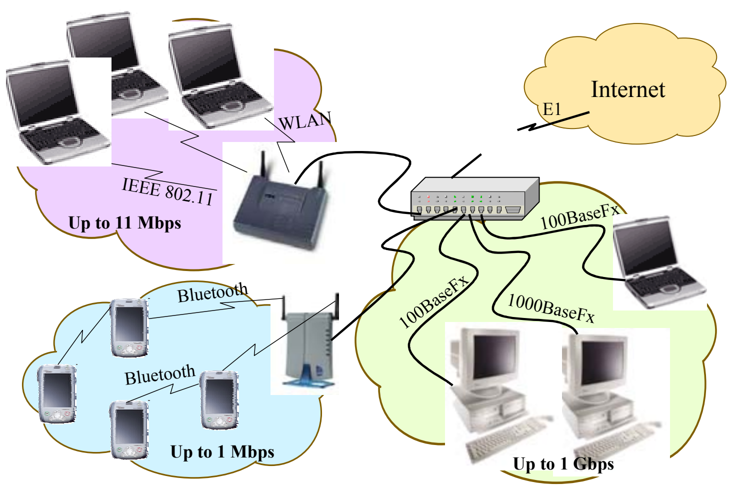


Figure 1.7 Local Environments with Heterogeneous Links and Capacities

The environment with heterogeneous bandwidth capacities, however, exists not only in the Internet. It is also a case in Intranet where several devices with different bandwidth capacities, for example palmtop, personal digital assistance, wireless notebook, and high performance computers, are connected one to each other and exchanging media. Another environment is ad-hoc network which consists of various wireless devices with diverse capabilities. Therefore the problem statement is also valid in such environments in addition to the Internet case.

Figure 1.7 shows local environments where various devices with heterogeneous bandwidth links and capacities may exist. A segment of the environment consists of notebooks in wireless LAN environment, in which the maximum bandwidth is 10 Mbps. Another segment is equipped with bluetooth technology that can take only up to 1 Mbps bandwidth. The latest segment is composed of computers and notebooks with 100baseFx and 1000baseFx, and hence, is capable of accommodating bandwidth capacities up to 1 Gbps.

For the multimedia applications streaming over the Internet the consequences of the heterogeneity of bandwidth capacities in access points to continents, countries, and clients, and the variety of traffic intensity are two-fold:

1. In terms of streaming media, there is no "one format fits for all clients". Streaming high quality media creates bottleneck at the access point whose low bandwidth capacity and, in turn, intensifies the number of lost packets and increasing delay. Streaming low quality media fits the clients with poor bandwidth but sacrifices the potentiality owned by clients with higher bandwidth. To promote fairness, media should be delivered in various format and size to conform the bandwidth capacities of clients.
2. In regard with processing the media to fit various bandwidth capacities two options are possible.
    a. The server does all the process. It has to maintain the information regarding the client bandwidths and to perform varieties of data transcoding processes to change the format of media to fit the various bandwidth requirements. However, this approach has two drawbacks:
        (1) Processing overload is unavoidable in server, in case there are many clients connecting at the same time.
        (2) It is not scalable in terms of increasing number clients since the server handles the overhead alone and therefore no load sharing strategy is possible to implement
    b. The media post-processing is performed at the best-selected intermediate peer(s). The selection is made on the basis of performance criteria such as latency, delay and bandwidth capacity. Additional mechanisms are needed for intermediate peer selection process and for the activation of post-processing code at intermediate peer. Another necessary mean is mechanism to track intermediate peers that agree to cooperate to share their resources

to perform remote code execution. This approach offers clear advantages:

(1) Processing overload can be distributed to several nodes

(2) It supports the notation of multicast communication. The post-processing can be done in several multicast node. Member of a multicast group can be classified based on the bandwidth the clients have. For example, clients with high bandwidth are grouped within a multicast region and the other multicast group handles clients with poor bandwidth.

(3) It supports scalability by enabling distributed processes. The explosion of clients can be managed through several peers and, hence, decreasing the possibility of overloaded processes.

1.4 The Emergence of Active Networks

The Internet is originally designed to support text communication. A problem in today's Internet infrastructure may occur when a streaming multimedia application is to take place. The information content of video and audio signals that contain moving or changing scenes may simply be too great for Internet clients with low bandwidth capacity if no adaptation is performed.

Replacing the Internet in the near future is almost impossible because it has been very widely used and has obtained very high world-wide acceptance. Changing the Internet infrastructure to support any services that are intended to be instantly available is also not recommended due to lengthy standardization process. Newly proposed architecture and concept such as DiffServ, IntServ and IPv6 give samples on how difficult and tiring to propose additional layer, function and services to the current Internet architecture. In short, Internet structure is too rigid to support flexible and dynamic demands of future applications if new strategy is not introduced and established.

Active networks philosophy is to by-pass requirements for standardization by introducing a kind of programmable networks. It was originally proposed by Tennenhouse at MIT [TEN96, TEN97] to increase flexibility of network device by adding small programs into the packet header. This program is intended to run on the fly on the network devices that the packet encounters. This way, new services can be installed and run on demand in, and do not need permanently exist in the network. Service provider can upload the program from the network that supports the service once the demand is over.

Active networks demonstrate how protocols and services are introduced without discontinuing system operations. Active networks (ANs) enable faster protocol innovation by making it easier to deploy new network protocols and services. These networks are "active" in two ways [TEN96]:

1. Routers and switches within the network can perform computations on user data flowing through them
2. Users can 'program' the network, by supplying their own programs to perform these computations

1.5 Hypothesis

The dissertation work proposes an improved active network concept and architecture for distributed and dynamic streaming multimedia environment with heterogeneous bandwidth capacities that requires no changes in router level management and put no additional requirement to the current Internet architecture. Active network philosophy that is adopted for the architecture allows transcoding of video and audio format to take place on-the-fly in the network. This technique provides the capability of adapting multimedia traffic to the bandwidth capacities of the peers or clients. Transcoding is the process to transform a video format to a more bandwidth efficient one. A sample of this approach, namely friendly active network system (FANS) has been designed and developed.

The architecture of FANS introduces tracking mechanism and announcement and update algorithm that make FANS capable of maintaining the dynamic existence of peers or FANS clients. Furthermore, these functionalities in addition to accounting algorithm enable FANS to decide whether it is necessary to perform transcoding process to the multimedia traffic at intermediate peer, considering that this additional process is expensive in terms of delay but advantageous in case of reaching clients with poor bandwidth capacities.

The improved concept and architecture incorporate also a user interface for dynamic multiple locations streaming multimedia environment. While FANS is intended to be capable of reaching the client with poor bandwidth, it is also imperatively important that the user interface of the system is simple, self-descriptive and encouraging for the users. The usability of a multimedia system like FANS depends significantly on the design of the user interface. That is, the audio-visual display and required actions or procedures that manage user interaction with the system. Computer system with poorly designed user interface discourages users from making use of it.

With all of these concepts, architecture and functionalities, and the factual requirement to provide QoS for multimedia application streaming over the current Internet condition and other environments with heterogeneous bandwidth capacities, the hypotheses of the dissertation are:

1. Heterogeneity of bandwidth capacities is the dominant factor that affects the QoS of streaming multimedia applications
2. Employing our improved active network concept and architecture Friendly Active Network (FANS) is capable of streaming multimedia applications over environments with heterogeneous bandwidth capacities
3. The improved Active Network Concept and Architecture is capable of:
    a. Improving the QoS of streaming applications by reducing the number of packet loss and increasing the traffic reliability to reach peers/clients with poor bandwidth
    b. Promoting fairness in terms of QoS vs Bandwidth
    c. Supporting mobility, scalability and dynamic existance of clients/peers
4. FANS requires no changes in current Internet structure and, hence, is implementable in the near future

1.6 Contributions

Considering the above aspects and with the intention of having more institutions from dispersed countries to collaborate in educational processes and research projects on the field of multimedia and software engineering contributions of the dissertation are as follows:

1. The provision of system concept and architecture for dynamic multiple locations streaming multimedia environment, which possesses comparative advantages against other similar systems:
    a. The system is capable of accommodating a range of client's bandwidth until as lowest as 100 Kbps for 12 MB media
        As a comparison, proposal by Yamada et al [YAM02] can accommodate as lowest as 2 Mbps for 8 MB media
    b. The system proposes no changes in current Internet architecture and, hence, applicable in the near future

c. The system introduces not only dynamic routing but transcoding capability as well

   Resilient Overlay Network (RON) [AND01] can be considered as a mere dynamic routing approach to the dynamic condition of The Internet

2. The proposed concept and architecture that successfully promotes fairness in terms of quality of service between peers with various bandwidth capacities. Peers with higher bandwidth capacity receive higher quality of multimedia applications than peers with lower bandwidth capacity.
3. The provision of intuitive and easy to use user interface for dynamic multiple locations streaming multimedia environment, which conforms the usability attributes for such environment
4. The provision of algorithm to selectively and dynamically involving the intermediate node's enhancement post processing by taking into account the size of multimedia application and bandwidth capacity of the clients Selectiveness is a necessary requirement due to the expensiveness of intermediate post-processing in terms of delay. FANS weighs up the suitability of involving intermediate post-processing dynamically at run-time and based on several performance parameters gathered during the session.
5. Dynamic tracking system and algorithm to support the notion of mobile and pervasive devices, and scalable and distributed systems.

1.7 Potential Applications

The development of learning technology has triggered evolution in learning environment. Time and place are the major constrains which create learning barriers. Aiming to overcome the time barrier, learning media has changed from classroom approach to TV or radio program, and through distributed CDs and video cassettes, and finally through mailing list, web-based teaching (any time). Dealing with barrier in terms of place, learning environment has also evolved from giving lecture at the same place (e.g., classroom) to at some places such as distance learning to some classes at different locations and, finally, to any place, for example virtual class such as distributed and mobile learning through Internet connection to mobile devices. This evolution is depicted in Table 1.2.

Ideal learning environment provides high quality education with widest accessibility and lowest cost. However this ideal condition is very difficult to accomplish. Therefore the more realistic principle is to choose two out of those three factors (quality, accessibility and cost). That is to provide either high quality education at low cost but limited access (e.g., in a multimedia room) or giving widest access to people at low cost but at the expense of quality (e.g., education through broadcast, static TV and radio program). The best but most expensive option is by ensuring high quality and widest access at very high cost, that is, a distributed multimedia learning environment through unlimited broadband access that reaches students everywhere and at anytime.

Table 1.2 Evolution in Learning Environment

Boundaries		**Media/Format**
Time	Same time	Classroom
	Some times	TV, Radio
	Any times	Distributed CD, Video Cassettes
Place	Same place	Classroom
	Some places	Distance learning, e.g Open University
	Any places	Virtual class (Distributed and mobile learning through the Internet)

Open universities, such as British Open University in UK and Indira Gandhi University in India are some examples of the strategies to extend the accessibility level of education at low cost. In sharing the course contents, most of the open universities throughout the world are mainly using traditional, paper-based method of learning. However, some open universities have started offering multimedia-based courses to students connected through Internet. The interactive and physical contact is maintained through the streaming audio and video, and shared electronic whiteboard displayed on the screen shared by all class participants.

Friendly Active Network System (FANS) can be implemented to address the issues on the problem of multimedia applications streaming through clients with heterogeneous bandwidth. This situation exists throughout the world where universities or institutes connected through high-speed bandwidth are expected to deliver multimedia teaching sessions to other universities or institutions with various bandwidth capacity. Delivering a high quality video creates congestion to institutions with poor bandwidth. On the contrary, delivering low quality of video underutilizes the institutions with high bandwidth capacity. There is no simply one quality fits for all bandwidth capacities.

FANS approach is to fairly distributing the quality of video presentation based on the capacity of bandwidth possessed by partner institutions. Institutions with high bandwidth capacity receive higher quality of video whereas institutions with low bandwidth capacity get lower quality of video.

By addressing the multimedia applications streaming through the environment with heterogeneous bandwidth FANS supports the distributed learning setting and access philosophies. The balance between quality, access and cost is accommodated based on the specific need of the partner institutions. Institutions with limited bandwidth due to financial constraints or other reasons can get equal access, although with lower quality, as the institutions whose higher bandwidth. Moreover, given its capability to accommodate device with lower bandwidth, FANS is potentially capable of supporting additional applications running on mobile device, given their universal resource locator (url) address is announced and, therefore, trackable by FANS tracking mechanism. It is a kind of giving lectures or multimedia sessions to any place anywhere the Internet exists. The user of mobile device such as wireless notebooks and personal digital assistants (PDAs) can connect to FANS once they are connected to the Internet.

Potential application for FANS also includes ad-hoc networks where the existence of application level active node could be used to adapt the media to the various bandwidth capacities and a variety of processing capability of ad-hoc devices. Further, given FANS capability of tracking peers dynamically and of gathering performance information at run-time, it is possible to extend its applications for pervasive environment where each device dynamically cooperates and coordinates to each other and is capable of performing communication through active gateway.

1.8 Publications

Ramli, K., Ekadiyanto, F.A, Hunger, A., "Utilizing ALAN Concept to Improve the Performance of Streaming Multimedia Applications over Heterogeneous Bandwidth Environment", to appear in *Springer's Lecture Notes on Computer Science: Proceedings of Worskshop on Active Network Technology and Applications*, Osaka, Japan, May, 2003, pp 129-140.

Ramli, K., Hunger, A., Erdani, Y., "A User Interface for Dynamic Multiple Locations Streaming Multimedia Environment", *Proceedings of IASTED Computer Science and Technology Conference*, Cancun, Mexico, May, 2003, pp. 348-352.

Ramli, K., Ekadiyanto, F.A, Hunger, A., "A Cooperative Distributed Computing Concept to Improve the Quality of Streaming Multimedia Applications over the Internet, *Proceedings of the ISCA 18th International Conference on Computers and Their Applications (CATA 2003)*, Honolulu, USA, March, 2003, pp. 417 – 420.

Ramli, K., Buhari, MIS., Hunger, A., "FANS Approach for Streaming Multimedia Applications", *Proceedings of International 7th International Student's Scientific Meeting*, October 2002, Berlin, Germany

Ramli, K., Buhari, MIS., Hunger, A., "On the Design, Development and Implementation of a Prototype of User-Friendly Active Network System (FANS)", *Proceedings of International Conference on Electrical, Electronics, Communication and Information (CECI)*, March, 2001, Jakarta, Indonesia

Ramli, K., Hunger, A., Buhari, MIS., "Towards the Development and Implementation of FANS: A User-Friendly Active Network System", *Proceedings of 5th International Students Scientific Meeting (ISSM)*, October 2000, Paris, France

Except for the paper for ANTA-2003 in Osaka and IASTED-2003 in Cancun, Mexico, other papers are accepted without comments or modification. APPENDIX - J lists the reviewer's comments on the Papers.

1.9 Dissertation Outline

Chapter Two discusses current efforts and technologies to address the evolution of the Internet to support newly developed services, including multimedia applications. The emergence of approaches that propose the modification of Internet structure and the utilization of intermediate node and cache is described. The improved active networks architecture inferred in this chapter is based on the recent developments in networking technology. The active networks (AN) and application level active networks

(ALAN) are explained in Chapter Three. In this chapter the reasons for the choice to adopt ALAN concept instead of original AN concept are outlined.

FANS design considerations and implementation is discussed in Chapter Four. After observing the challenges outlined in Chapter One, putting some considerations related to the objective of the work, and gathering ideas and knowledge from other works in the field as explained in Chapter Three and Chapter Four, the design and implementation of the proposed solution to the obstacles faced by multimedia applications to stream over the environment with heterogeneous bandwidth without having to modify the current Internet structure are described. In Chapter Five the quality of service (QoS) metrics that could be used to measure the advantages offered by the proposed mechanism are identified and the experimentation testbed to emulate the dynamic nature of the real Internet is explained. The results of performance measurement are presented and discussed in Chapter Six. Chapter Seven concludes the dissertation and outlines some directions and issues for future works.

Chapter Two

Facilitating the Evolution of the Internet to Support Emerging Multimedia Services

2.1 Introduction

Current and future Internet applications are most likely, having to cover streaming multimedia applications. A new generation of distributed applications, such as teleteaching, telemedicine, and other cooperative groupworks, are also being deployed in Internet. These applications are demanded to deliver adaptive and satisfactory quality of service (QoS), in order to be accepted by general users

Research works in the field of streaming multimedia applications and their scalability issues can be categorized into three main subjects: quality of service (QoS) for streaming media, QoS-aware middleware, and media or service discovery and selection. These fields can generally be classified as shown in Figure 2.1.

There are many ways to represent QoS. One might argue that bandwidth is the point. However, with the various need of bandwidth it is very hard to determine when and what kind of bandwidth that is sufficient to represent QoS. Sometimes, low bandwidth is enough, for example for text transfer. It is also often high bandwidth means nothing if the counterpart could not support it or could not communicate with comparable bandwidth. It is the latest condition that we are now facing in implementing multimedia applications to stream over the environment with heterogeneous bandwidth like the Internet.

The user‘s level QoS requirement must be mapped into lower level metrics. This is the function of QoS-aware middleware. QoS-aware middleware is an important layer if the scalability of multimedia system and heterogeneity of receiver‘s bandwidth are of this work’s concerned. It accepts QoS specification from upper level and transforms the specifications into parameters such as bandwidth, delay and jitter. In Qos-aware middleware terminology, these processes are known as QoS translation and compilation. QoS-aware middleware components, described in sub-section 2.5.2, include resource monitoring, resource broker and traffic adaptor. Network interface is responsible to forward streaming data and to perform QoS setup and enforcement. Its action is based on instruction dictated by QoS-aware middleware.

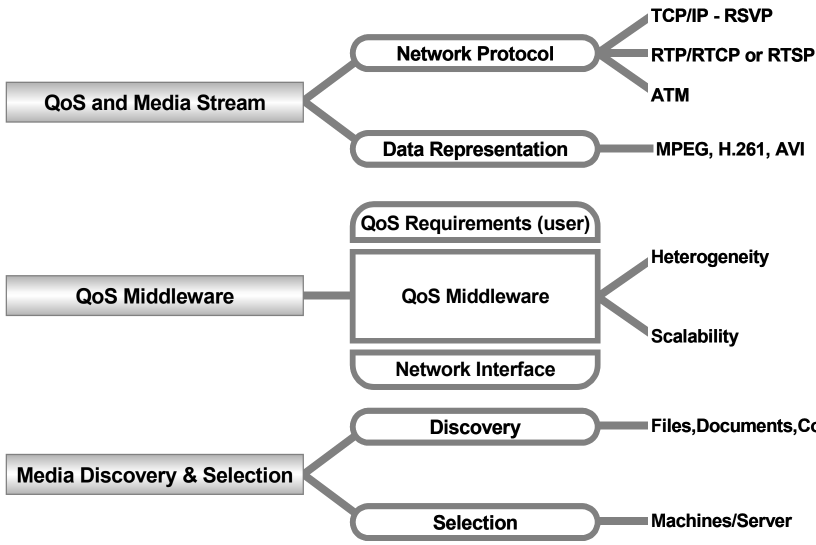


Figure 2.1 Skeleton of QoS and Scalability Issues

Several protocols to improve the Internet protocol have been suggested. Real-Time Protocol (RTP), for example, has been deployed to complement with TCP/IP, UDP/IP or other protocols in dealing with multipoint multimedia communication. Resource reservation protocol (RSVP) [ISI99], yet another QoS-oriented protocol, is suitable for client to ask all nodes along the path between server and client to reserve a certain amount of resources for the client in order to get the level of quality of service it requires. The RSVP diagrammatic function is shown in Figure 2.2. Asynchronous Transfer mode (ATM) [CIS95] is appropriate for high-speed backbone that works based on fixed cell and statistical multiplexing access scheme.

Discovery is the process to find files, documents, or code in the global Internet. Selection means to select, based on some performance metrics, the “best‟ machine or server among a group of machines that provide the same service. Services here could be a trancoding code or enhancement protocol such as a Transport Protocol for Real Time Applications (RTP) or Real Time Streaming Protocol (RTSP) [RFC2326]. The process involved in selecting best server and performing enhancement process is referred to as QoS adaptation.

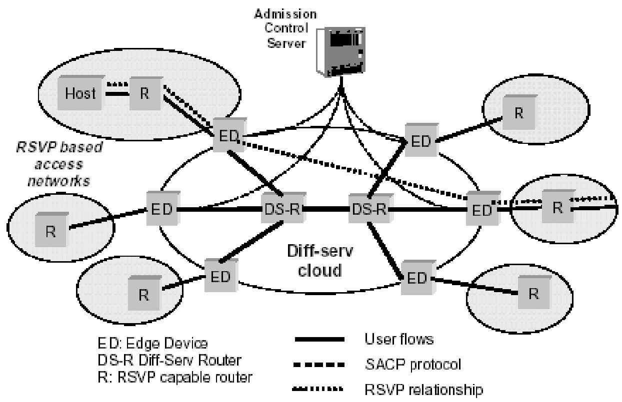


Figure 2.2 RSVP Diagram Sample [DET00]

Scalability is another important issue for the future multimedia streaming applications and other network computing systems. The proliferation of the Internet, the availability of powerful computers and high-speed networks contributes to the scalability problem. To be widely used, a system must support and anticipate a growing number of users or device connected to it.

In the field of multimedia services scalability can be accomplished by the combination work of QoS-aware middleware and media discovery service and selection. Selection is a procedure to detect and locate particular machines that are willing to cooperate and to check the availability of code or protocol to be utilized to improve network performance. For active network technology, explained in Chapter Three, the process of dynamic discovery and selection is required in selecting intermediate peers to perform enhancement services. This is a kind of directory services in service selection mechanism. The intermediate peers then should be able to decide whether or not the current network condition forces them to perform enhancement process on the data streaming over them. Discovery and selection mechanism can be extended further to support users with mobility.

Cooperation between machines is also a key factor and has been widely known as computer supported collaborative work (CSCW) or distributed system. In a

QoS-supported network machines contribute their resources for common purpose. The resources may be information regarding network condition around the machine, cache, or active code resides in the cache. The data streaming over the machine can utilize the code if necessary. The available cache can be used to temporarily store active codes. Java byte capable of executing at run-time with the help of Java class loader is an example of active code.

2.2 Providing QoS for Media streaming Over the Internet

The Internet has recently been experiencing an explosive growth of the use of streaming audio and video. Such applications are delay-sensitive, semi-reliable and rate-based. Therefore assurance of the end-to-end quality of service (Qos) for the increased traffic will be essential for these particular Internet-based services. Quality of service itself cannot be simply defined because it means many things to many people. However the central idea of the QoS is the provision of network resources that an application requires to satisfactorily work.

2.2.1 Integrated Services and Differentiated Services

Current Internet infrastructure does not provide mechanisms to ensure an upper bound on end-to-end delay or a lower bound on available bandwidth. Consequently, the quality of delivered services is neither controllable nor predictable. However, lack of support for QoS has not prevented rapid growth of realtime streaming multimedia applications and this trend is expected to continue.

Internet engineering task force (IETF) is evolving QoS support mechanisms for the next generation Internet services. Two approaches that are already on the development phase are the integrated services (IntServ) [RFC2215] and differentiated services (DiffServ) [RFC2475].

The IntServ watches over the QoS of the individual microflows to apply policing strategies. However, since IntServ retains state of each microflows, it is too complex to manage for large networks due to the potential state explosion and will not scale easily.

DiffServ architecture, on the contrary, was conceived by IETF as a way to provide a scalable service differentiation over the Internet by aggregating the behaviour of the individual microflows. However since DiffServ works without reservations, local congestions and violations of the service parameters cannot be prevented. A sample of DiffServ capable network is presented in Figure 2.3.

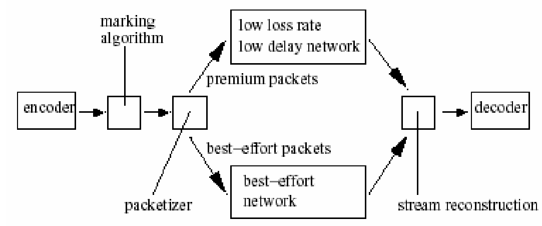


Figure 2.3 A Block Diagram of DiffServ Network Architecture [VIT02]

Two-tier resource management model [TER99a, TER99b] for the Internet is orientated to achieve scalability and flexibility by making fundamental distinction between the two components of architecture: forwarding path and management plane. The low level forwarding paths provide different level of network services. It consists of a bandwidth broker and a policy database. Bandwidth broker is responsible to configure routers forwarding parameters, whereas policy database stores information about flows requiring increased network service. Management plane carries out configuration of network nodes with respect to which packets get special treatment and what kind of rules are to be applied to the use of resources. DiffServ service control element is responsible to enforce that traffic conforms to predefined profiles. Policers cut off excess traffic and sharpers provide temporary buffering to make traffic conform to the specified profile. However, since it is not easy to do experimentation for the two-tier model in wide area network, IRLSim [TER00] is deployed as simplification tool. IRLSim is therefore only a software simulator based on the concept of diffserv.

A DiffServ experiment [TER99a] shows that this approach works with high precision to control expedited forwarding (EF) traffic and best effort (BE) traffic. EF has higher priority than BE. If EF occupied x% of the bandwidth capacity then BE would use the rest (100-x)%. DiffServ puts upper-bound limit for EF traffic and when it is not fully used, BE traffic ramps up and consumes the extra available bandwidth.

Nonetheless, both DiffServ and IntServ approaches are arguably contradictory to one of the significant Internet philosophies, which is to keep complexity to the network edge. This important philosophy is considered thoroughly in the proposed system and architecture.

2.2.2 Layered Multicast

The Internet's bandwidth heterogeneity and scale makes streaming multimedia communication design even more difficult. For streaming multimedia, one would like to broadcast or multicast live, streaming data from any particular sender to arbitrary large users connected to Internet with potentially high variability in bandwidth capacity. The simplest solution to this problem is to distribute a uniform representation of signal to all interested users using IP multicast. Unfortunately, with this approach, the clients with low-capacity bandwidth might suffer congestion while high-capacity regions are underutilized. Heterogeneity in bandwidth causes other rich kinds of heterogeneity in terms of packet loss, mean delay, delay jitter, and maximum delay.

One of the proposed solutions to overcome the problem is layered multicast [GUO01, SHA92, TAU94]. The streaming data are subdivided into a hierarchy of cumulative layers and each of the layers is transmitted to a unique multicast address. A signal is encoded into a number of layers that can be incrementally combined to make gradual refinement. Heterogeneity is achieved by locally degrading the quality of the transmitted signal by dropping layers at certain points in the network. A sample of layered multicast approach for video over Internet is shown in Figure 2.4.

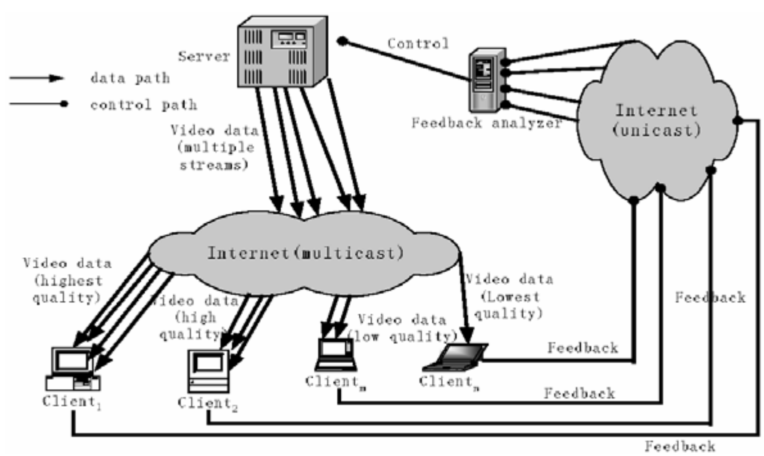


Figure 2.4 Architecture of Layered Multicast for Video over Internet [GUO01]

These early works are further followed by the deployment of a receiver-driven layered multicast [CAN96] in which the different layers of the hierarchical signal are reorganized across multiple multicast groups. Receivers adapt to congestion by adding and dropping layers by joining and leaving multicast group. The authors also designed and simulated an experimental network protocol called Receiver-driven Layered Multicast (RLM). Under RLM, multicast receivers adapt to both the static heterogeneity of link BW as well as dynamic variations in network capacity (i.e. congestion).

Authors in [GUO01] set up simulation experiments with 100 receivers in total. The receivers belong to 5 bandwidth categories, ranging from 64 kbps to 2048 kbps. There are 5, 5, 20, 30, 40 users corresponding to the categories, respectively. In each category, the bandwidth probability density functions of all receivers follow the same Gaussian distribution. The mean values of the distribution vary from time to time, although the variances of distribution are fixed. In each simulation, they change the means of the Gaussian. They conclude that the fewer the number of layers, the more the improvements. This is because when the number of the layers increases, the rates of each layer can fit individual receivers better even if the rates are evenly located. In addition, the higher the bandwidth dynamics of each user category, which means the closer the overall user distribution to a uniform shape, the less the improvements.

Protocols that use layered multicast groups, however, provide no feedback to the sender and only allow the receivers to participate in congestion control by selecting an appropriate set of the layered groups. These feedback-free schemes, consequently, have the following two disadvantages [GOR1]. Firstly, excluding the sender from the congestion control loop weakens the ability to protect against receiver misbehaviour. Secondly, these protocols suffer from a significant mismatch between the statically allocated transmission rates of the groups and the changing capabilities of the heterogeneous receivers

2.2.3 Media Gateway

A primary purpose of the media gateway is to regulate the output bandwidth of the transcoded packet stream. Video gateway, for example, accomplishes bandwidth adaptation through transcoding and rate control. It generates variable rate output and, if necessary, dropping frames to meet a given rate constraint. Transcoding is the process by which the gateway transforms a compressed video bit stream into a different bit stream, usually the lower one, either by employing the same compression format with alternate parameters or by employing another format altogether. Media gateway approach, however, is similar to router level active network. That is, it requires the router level mechanism to be adjusted and modified.

Application level video gateway [AMI95] incorporates RTP as an integral component. The implementation supports only JPEG/H.261 transcoding model. Streams, which are not JPEG, or are not intended to be transcoded are simply forwarded across the gateway. The purpose of the scheme is for multicast environment. Application level video gateway provides for matching the transmission quality to the heterogeneous bandwidth constraints of distinct regions of a single logical multicast session. The gateway does so by intelligently managing incoming and outgoing video streams using transcoding and rate control. Nonetheless modifying the rate control for video application is not a wise option since it can affect the smoothness motion of video pictures, which is a more important factor of a good video than justifying the quality of picture to lower one, for example by transcoding JPEG to H263.

An RTP to Hyper Text Transfer Protocol (HTTP) video gateway software [JOH01] enables Internet user to take part of multicast video streams, despite at potentially high latency and low frame-rate. The only pre-requisite is the access to the WWW through standard browser. WebSmile is a software component installed on an ordinary Web-server to give users access to multicast RTP video streams through the Web-server using HTTP streaming. Assuming that the bandwidth bottlenecks are the HTTP connection rather than the multicast backbone, information about the BW constraints of the HTTP connections is used as input to the multicast flow control algorithm. TCP drives the decision algorithm for subscribing to multicast layers. This is known as the TCP-driven multicast flow control algorithm. This work gives an idea of bridging unfriendly protocol for streaming applications like HTTP with a generally accepted streaming protocol such as RTP.

Degas [OOI00a] is a programmable, application-level media gateway. It allows user to “inject‟ user-defined program, called deglets, into a gateway to perform customized transcoding, filtering and mixing of video and audio streams of a multicast session. The work enables a video-enhanced Web and uses Vosaic as their WWW browser. As part of Degas architecture, Video Datagram Protocol (VDP) is introduced. VDP is specialized for handling real-time video over the WWW. VDP reduces inter-frame jitter and dynamically adapts to the client CPU load and network congestion. However, Vosaic project in general can be considered as merely creating video enhanced web system.

Performance of Degas is observed in [OOI00a]. They run experiments to understand the overhead introduced by the gateway. Video streams were sent using vic [CAN95] from hosts connected to the gateway using an 100 MB Ethernet. Receivers are located at the same LAN. They ran experiment to measure the overhead introduced by optimizer and the savings caused by the optimisation. A 352 x 288 H261 video stream at 8 fps and a motion JPEG video stream of size 176 x 144 are used. For the purpose of the experiment they develop the optimiser routing for rescaling of video frame. Their experiment confirmed that the timing overhead in optimizing is small (<1 ms), while the savings are significant (about 150%).

2.2.4 Resilient Overlay Network

Resilient Overlay Network (RON) [AND01a, AND01b] is a distributed Internet architecture that is capable of detecting and recovering from path outages and periods of degraded performance within several seconds, improving over today's wide-area routing protocols that take at least several minutes to recover. A RON is an application layer overlay on top of the existing Internet routing substrate. The RON nodes monitor the functioning and quality of the Internet paths among themselves, and use this information to decide wether to route packets directly over the Internet or by way of other RON nodes, and hence optimising application-specific routing metrics.

The main goal of RON is to enable a group of nodes to communicate with each other against problems with the underlying Internet paths connecting them. RON recognizes problem by aggressively probing and monitoring the paths connecting the nodes. RON nodes exchange information about the quality of the paths among themselves via a routing protocol and build forwarding tables based on a variety of path metrics such as latency, packet loss rate, and available throughput. Each RON node obtains the path metrics using a combination of active probing experiments and passive observations of on-going data transfers. In short, RON supports monitoring-based selective routing. Figure 2.5 depicts a sample of RON architecture in implementation.

RON people measure the 30-minute average packet loss rates between MIT and ArosNet. In these samples, the loss rate between MIT and ArosNet ranged up to 30%, but RON was able to correct this loss rate to well below 10% by routing data through Utah (and occasionally through the cable modem site).

This shows that situations of non-transitive Internet routing do occur in practice, and can be leveraged by a RON to improve the reliability of end-to-end application communication.

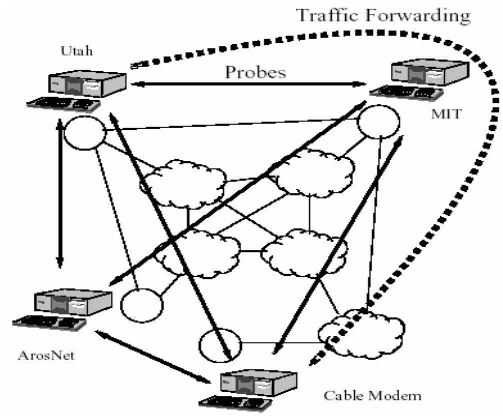


Figure 2.5 A General Approach Used In RON System [AND01b]

In another experiment the latency results between MIT and ArosNet, are gathered over 62 hours between January 8 and January 11 2001. In 92% of the samples, the latency of the packets sent over a RON-like path was better than the Internet latency. The average latency over the measurement period decreased from 97 ms to 68 ms.

RON’s aim is to improve applications performance over the Internet by providing alternative routing to the Internet routing in case of resource drop. RON, however, requires additional aggressive probing that could affect the performance of Internet due to overhead process. In addition, no mechanism is provided to adapt bandwidth-consuming data to environments with limited bandwidth capacity.

Providing QoS for media streaming over heterogeneous bandwidth environment such as the Internet is a complex job and in most cases, the proposed solution are application specific. Each of the above concepts and approaches and present are compared and summarized in Table 2.1.

Learning the advantages and drawbacks from the above concepts and approaches the proposed concept and architecture for streaming multimedia environments with heterogeneous bandwidth imposes the following characteristics:

1. Demand no modification to the current Internet structure, in particular it should not increase the router level complexity
2. Keep the complexity to the network edge, or even to the application level. This philosophy is in accordance with the Internet paradigm
3. Support for scalability as the number of system's participant might grows significantly
4. Extend the approach taken by RON to include active processing in the middle node like what has been proposed by media gateway.

Table 2.1 Concepts and Approaches in Streaming Media over the Internet

Concept and Approach	Advantage(s)	Disadvantage(s)
Resource Reservation Protocol (RSVP)/ Integrated Services	• Qos Support to the application	• Increase router complexity • Lack of scalability support
Differentiated Services	• Support for scalability and traffic conditioning • Low overhead to router	• Increase gateway complexity • No local congestion prevention
Layered Multicast	• Support clients with heterogeneous bandwidth	• Increase router complexity • Static allocated transmission rate due to no receiver feedback
Media Gateway	• Media delivery with adjustable bandwidth capability	• Increase router level complexity • No scalability potential
Resilient Overlay Network	• Support for congestion avoidance and recovery • Improve traffic reliability	• Merely a rerouting policy • Not addressing clients with bandwidth heterogeneity

2.3 Protocols for QoS-Supported Media Delivery

In many cases protocols are application specific. Although the Internet is originally designed for text transfer, it currently is the most widely used protocol and its applications are broadened. To be widely accepted and deployed, newly emerged protocols and network applications should at least support internetworking with the Internet protocol or work on top of it.

Transfer Control Protocol (TCP) [RFC793] is a reliable, connection-oriented packet delivery protocol used for data transfers by Web browsers and their associated Hyper Text Transfer Protocol (HTTP) servers. For continuous media, such as real-time video, full file transfer mechanism and TCP are nevertheless unsuitable due to several reasons [CHE95]:

- TCP imposes its own flow control and windowing schemes on the data stream. These mechanisms effectively destroy the temporal relations shared between video frames and audio packets
- TCP cannot support multicast which is one of the effective way to distribute streaming video and audio
- Reliable message delivery is not required for video and audio. Video and audio streams tolerate frame losses. Losses are seldom fatal although detrimental to picture and sound quality. Retransmission of TCP causes jitter and skew internally between frames and externally between video and audio streams.

User datagram protocol (UDP) [RFC768] is the alternative to TCP. RealPlayer, StreamWorks, and VDOLive use this scheme. UDP eliminates TCP's error correction and allows packets to drop out if they are late or damaged. Due to lower requirement for reliability factor of its data transmission, UDP-like protocol is arguably better for continuous media delivery. However, an obstacle for UDP is the fact that many Internet hosts are located behind firewalls. In the general case firewalls do not allow UDP-based real-time traffic to pass through and in many cases they also employ techniques like network address translation (NAT) that complicate end-to-end real-time communication.

Transport Protocol for Real Time Applications (RTP) [RFC1889] is an application-level protocol that is designed to conform to the needs of multiparty multimedia communications and applications. RTP contains no specific assumptions about the capabilities of the lower layers, except that they provide framing. It runs on end systems and provides demultiplexing capabilities. The protocol consists of two parts: the data transfer protocol RTP and the RTP control protocol or RTCP. RTCP provides mechanisms for data distribution monitoring, cross-media synchronization, and sender identification. RTP has no protocol state by itself and can thus be used over either connectionless networks, such as IP/UDP, or connection-oriented networks, such as XTP or ATM (AAL 3/4 or AAL5).

Video Datagram Protocol (VDP) [CHE95] is an improved RTP, i.e., RTP with demand resend. On top of it, Vosaic [CHE96], short for Video Mosaic was developed. Vosaic extends the architecture of traditional WWW browser by incorporating real-time video and audio into standard hypertext page, which are displayed in place. The design of Video Datagram Protocol (VDP) is targeted to make efficient use of the available network bandwidth and CPU capacity for video processing. It is not same as RTP in that it takes advantage of the point-to-point connection between Web server and Web client. VDP uses an adaptation algorithm to find optimal transfer bandwidth. A demand resend algorithm handles frame losses. When configured in Java, this protocol, like HTTP is invisible to the network and can stream through firewalls.

Adaptive Gateway Location Protocol (AGLP) [OOI00b] is used in Degas for choosing a gateway that efficiently utilizes bandwidth. It is a soft-state protocol based on the announce-listen model already widely used in MBone. AGLP adapts to changing network conditions as well as the dynamic existence of gateways, senders and receivers by migrating computations, also called services, between gateways. It assigns a new service to a gateway rapidly.

Considering the above description, the existing multimedia orientated protocols already standardized and widely accepted, and to accommodate the potential applications and intended users of the system, the following protocols are selected to be embedded into FANS:

1. RTP/UDP as the network protocol
   RTP is an application level protocol that is independent of lower level protocol. Moreover, it is capable of supplying feedback on end-to-end delay, number of lost packet and number packet duplication. Running RTP over UDP offers the the same level of acceptance as the commercial products such as RealPlayer and StremWorks. UDP is better for continuous media than TCP.
2. HTTP as the protocol for tracking peers joining and leaving the multimedia session.
   A main factor of this selection is due to the popularity of HTTP protocol among the Internet (browser) communities. This contributes to the simplicity factor of the system before the users. Additionally firewall generally allows HTTP protocol to go through the Internet host and, hence, eliminate the UDP blocking cases.

2.4 Issues on Scalability and Distributed Resources

The scale of distributed system like the Internet grows very quickly and its traffic continues to increase at an exponential rate. The development of the network infrastructure, on the other hand, cannot follow this speed. As a consequence, the network has become the bottleneck of the distributed system. A good example is the World-Wide Web. Suppose a popular web source has an extremely burst of access at a special time, the server load can be significantly increased at that moment. Therefore to scale a distributed system more efficiently, the use of cache on the client sites to store data or having intermediate repository for popular media is very useful to reduce the network bandwidth consumption and server load.

The dynamic nature of the Internet prevents us further from solving the problems of link and server overload, network bottleneck and scalability using a centralized approach. Recent development in the evolution of network technology, such as the global scale of interconnection and the availability of high-speed broadband networks, have forced anybody to consider network resources as components of a global distributed system. Instead of running on a single computer, large-scale systems execute on a number of different machines that might locate at geographically dispersed locations. A few simple reasons are to distribute processing and to optimise local and network resource utilization through resource sharing.

2.4.1 Service Location Protocol

A problem that arises in a large-scale distributed system is to locate server and object. Shortening user-perceived latency in an Internet is valuable for user satisfaction. A way to reduce user-perceive latency is through increasing the probability to find a service faster.

The Service Location Protocol (SLP) working group of IETF [RFC2165, RFC3111] has been developing protocols to provide a scalable framework for the discovery and selection of network services. However, It is not a global resolution system for the entire Internet. It is, more exactly, intended to serve enterprise networks with shared services. One of the goals is to support dynamic discovery services in new environments.

2.4.2 Service Replication and Dynamic Server Selection

The increasing popularity of distributed information services like the WWW has resulted a clear evident of problems of scale. Three impediments to the centralized information services that trigger the attractiveness of such distributed information services are [CAR97]:

- Excessive server load due to popular documents repeatedly retrieved by client
- Wasted network bandwidth due redundant document transfer
- Excessive latency due to the potential transfer over slow paths

Service replication [FEI98] can be considered as way to improve the ability of a service to manage a large number of clients. Service replication is a form of distributed database system maintaining a number of copies of popular document or code. An important function of service replication is its ability to redirect client requests to the best server, according to some optimality criteria. The goal is to allocate servers to clients in a way that minimizes a client's response time.

The main obstacle for a good solution to replication policy is the heterogeneity of documents, may it be size, popularity, geographical location, and frequency of updates. Differentiated strategy for replicating web documents [PIE00] proposed that each document be replicated with a policy specifically tailored to it. Their approach shows a significant performance improvement with respect to end-user delays, wide-are network traffic and document consistency.

Employing dynamic server or service selection [CAR97] mechanism instead static assignment enables application-level congestion avoidance. Dynamic server selection provides technique for finding good service providers without a priori knowledge of server location on the network topology. Dynamic server or service selection outperforms static policies by as much as 50%.

2.4.3 Application Layer Anycasting

Complementing with the service replication idea, application layer anycasting [BHA97] examines the definition and support of anycasting at the application layer, providing a service that maps anycast domain names into one or more IP addresses using anycast resolvers. It is independent from network-layer support and includes the notion of filters, i.e., functions that are applied to groups of addresses to affect the selection process.

IP anycast address is used to define a group of servers that provide the same service. This approach is useful to determine the best server among similar ones. Anycast domain name (ADN) uniquely identifies a collection of IP addresses, which constitutes an anycast group. Request from user is forwarded to the best server, based on some optimality criteria among the group.

On their experiment they set three hosts on different subnets within local campus network and two machines on different subnets of the remote campus' network. The proxies on the local campus agree more often and more accurately with the base than the proxies on the remote campus. Even the proxies on the remote campus give fairly good results, selecting the best or second best server about 50% of the time. Compare this result with a random selection of servers –as might be obtained without anycasting support- would pick the best server 14% of the time.

The application layer anycasting is not used due to their stringent requirement for additional network address translator (NAT) and lookup table at certain nodes at the edge network. This approach, however, is considered for further FANS enhancement, in particular in addressing load balancing issues among FANS peers.

2.4.4 Grid Computing

Grid computing [BUY00, BUY01a] enables aggregation of distributed resources for solving large-scale data intensive problems in science, engineering and commerce. The real and specific problem that underlies the Grid concept is coordinated resource sharing and problem solving in dynamic, multi-institutional virtual organizations.

A set of individuals and/or institutions agrees to or bound to a certain sharing rules is referred to as virtual organization [FOS01, BUY01b]. The sharing are concerned not only with primarily file exchange but also to directly access to computers, software, data, and other resources, as is required by a range of collaborative problem-solving and resource-brokering strategies emerging in industry, science and engineering. This kind of sharing is, necessarily, highly controlled. Resource providers and consumers defining clearly and carefully just what are shared, who are allowed to share, and under which conditions sharing occurs.

A service discovery service records the identity and essential characteristics of "services" that are available to community members. A superscheduler routes computational requests to the "best" available computer in a Grid containing multiple high-end computers, where "best" can covers issues of architecture, installed software, performance, availability and policy. A replica selection service within data grid responds to request for the "best" copy of files that are replicated on multiple storage systems.

The concept of virtual organization from grid computing paradigm is very interesting and useful. This notion represents a group of peers or computers that agree to share their resources for common benefit. For example, to improve the performance of peers or to support the parallel computation works. This notion is adopted in FANS concept and architecture.

2.4.5 Content Distribution Network

An objective of content distribution networks (CDN) [STA00] development is to determine the most efficient mix of networking resources and storage for the delivery of web content and more recently streaming media. CDN views the network as a collection of devices that optimises the delivery of Internet content such as HTML documents and JPEG files by:

- caching contents near clients
- pushing content into those caches
- rerouting each client request to the best device available at that moment to serve the particular content and for the purpose of load balancing.

In a caching application, authors in [AGR01] modelled transparent surrogate architecture and translucent surrogate architecture. In transparent surrogate architecture, when a surrogate fails to satisfy a request, it contacts the origin server, gets appropriate data, and then serves the client request. In translucent surrogate architecture, a surrogate redirects clients to the origin server if it is unable to satisfy their request. It is intuitive to expect that putting more surrogates closer to the client population will decrease the response time experienced by the clients. However, this incurs an overhead necessary to maintain coordination between them. They conclude their experiments that the performance of content distribution network increases as the cache-hit rate increases. By increasing cache-hit rate from 50% to 90%, response time ratio decreases by 0.15 – 0.45 depending on the critical ratio.

The concept and architecture takes on the idea from this section to deal with scalability and mobility issues as follows:

1. Service discovery mechanism to find peer computers joining and leaving multimedia sessions
2. Dynamic server/peer selection to select the best intermediate peer to perform media post-processing
3. Content distribution network approach to allow remote peer to activate code resides in peer computers
4. Virtual organization concept of grid computing to represent a group of peer computers or institutions that are agree to share resources for the common benefit.

2.5 Guaranteeing Multimedia QoS in the Internet

With the widespread proliferation of Web services, quality of service (QoS) becomes a significant factor in distinguishing the success of service providers. QoS determines the service usability and utility, both of which influence the acceptability of the service.

QoS is a broadly used term that refers to a set of perceivable, performance attributes expressed to observe user-friendliness of a system with parameters that may be subjective or objective. Objective values are parameters related to a particular service such as delay, jitter and loss. These parameters are measurable and verifiable. Subjective values are based on the opinions of the end-users. In the later case, usually user-perceived latency or satisfaction is measured by means of questionnaire. The particular elements of a QoS definition depend on the information being transported.

2.5.1 QoS Measurement

Typical application QoS parameters for images and video include, but not limited to, image size, frame rate, start up delay, user-perceived latency and reliability. Large-scale multimedia streaming applications that are currently being employed possess significant problems as follows:

- User dissatisfaction due to poor QoS
- Poor cost-performance ratio due to inefficient management of system resources, in particular when guaranteed service is desired

Many metrics can be measured for QoS, since each application has its own view on QoS definition. Among them are the following:

1. Distance
   [GUY95] focused on distance measures with the objective of keeping communication local and limiting the use of long-haul links. [GWE94] proposed a selective server assignment based on geographical distance to place replicas near sources of high demand. Two obvious metrics for measuring distance in Internet are hops and round-trip latency.
2. User-perceived Response Time
   This is the time spent by user from initiating a retrieval process until getting the appropriate feedback from network. Appropriate feedback might be in the form of video-audio streaming or document. So, this is directly correlated with a user’s perception of the QoS. It is, however, a very difficult metric to monitor since it depends on server

capabilities (e.g., speed, numbers of processors at the server), current server load (e.g., number of queries currently being served), network path characteristics (e.g., propagation delay on the path) and current path load.

3. Frame Lost or Damage

   This affects the quality of pictures presented. Media streaming tolerate frame loss to some limit but not the delay. Delay makes the moving pictures breaking and user dissatisfaction of having these interrupted movies.

To verify the performance of FANS concept and architecture two of the three aspects listed above, i.e. user-perceived response time and frame damage or lost, are considered. It is claimed that in the current Internet structure distance plays insignificant role in affecting the quality of service. Observation of this work, regarding Internet traffic and redirection flow described in Chapter One, supports this claim. It is the bandwidth capacity and traffic intensity that count and influence the quality of service.

2.5.2 QoS-aware Middleware for Heterogeneous Environment

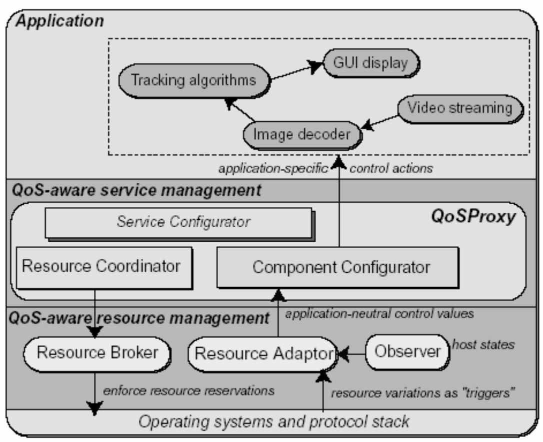


Figure 2.6 Components in QoS-Aware Middleware Architecture [NAH01]

A middleware architecture proposed by [NAH01] views a collection of interconnected application components on a single host as a set of tasks, with input-output dependencies. The architecture is depicted in Figure 2.6 and consists of two levels: QoS-aware resource management level and QoS-aware service management level. QoS-aware management consists of resource brokers, resource adaptors, and observers. QoS-aware service management is represented by a collection of middleware components, collectively referred to as the QoS Proxy.

The application component model defined in the framework enables the provision of end-to-end application QoS via QoS-aware middleware systems. With the experience from middleware system such as 2KQ[NAH00] and Agilos[LIB99] two important recommendations are outlined:

1. Resource-level QoS support via resource brokers, such as CPU and bandwidth broker, are highly desirable for the provision of end-to-end application QoS
2. QoS adaptation capability is necessary in middleware system, especially if they are to assist applications on top of best OS and networks.

2.5.3 Active Middleware to Control QoS Level of Multimedia Services

Service provision with negotiated and controlled QoS over best-effort network calls for a support infrastructure that activates intermediate nodes along the path between clients and servers. Multimedia Active Service for QoS (MASQ) [BEL01] is an active middleware solution for the QoS management of Video On-Demand streaming. MASQ exploits code mobility to establish an active path between the requesting client and the VoD server to customize VoD flows based on user profiles and device properties. MASQ then dynamically controls the offered QoS level to adapt locally when and where the network resource availability changes. The modular architecture of MASQ is depicted in Figure 2.7.

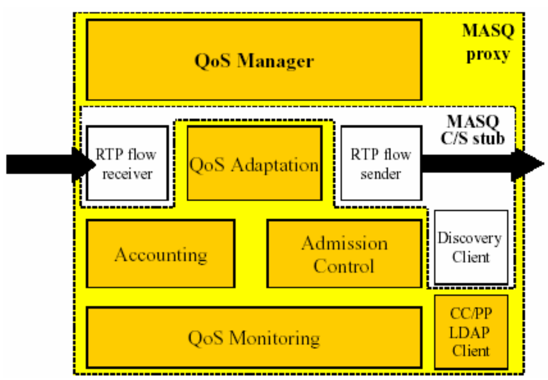


Figure 2.7 The MASQ Modular Architecture [BEL01]

QoS awarenes is a key requirement for VoD services over best-effort networks and QoS visibility is the property that an active middleware should be built around, to customize and to adjust dynamically provided QoS levels. Therefore, QoS customizing is done at negotiation time and QoS adaptation is performed at provision time.

FANS concept and architecture is combination of the technologies described in this chapter. Qos-aware and active middlewares in particular give me a foundation in developing FANS architecture. This is because improving QoS awareness and adaptation is among FANS significant targets. Therefore FANS system and architecture would be equipped with the following components:

1. Active environment for resource adaptor and QoS adaptation
2. QoS-aware resource management to provide information required in deciding appropriate action to adapt to the available resources
3. QoS monitoring to track the available resources and dynamic changes
4. Discovery and tracking mechanism to detect newly leaving and joining peers or clients
5. Dynamic peer selection based on the dynamic server selection approach. This approach has been proven to outperform static policies as much as 50% [CAR97]
6. RTP flow transmitter and receiver for streaming protocol

Chapter Three

Active Networks and Application Level Active Networks

3.1 Introduction

Active networks (AN) was originally proposed by Tennenhouse at MIT [TEN96, TEN97] to increase flexibility of network device by adding small programs into the packet header. This program is intended to run on the fly on the network devices that the packet encounters. This is generally known as the capsule approach. Another technique is utilizing intelligent node that is referred to as active node. The packets do not need to bring all fractions of codes. Instead, the packets contain a reference to a code, which already resides in the node.

There are a number of problems related to original Active Networks proposal [MAR99] and many believe that router level active networks are impractical. Application Level Active Networks (ALAN) [FRY98, PIA99] offer a more realistic and promising approach, or at least possess the potential as intermediate solution to router level active networks.

Note that it is very difficult to obtain performance result of projects in the field of active networks. Most of the early projects are intended to explore concepts, frameworks, languages and architectures of the active networks technology. This is also due to people of active networks are still even searching what the “killer applications” possible in this field.

3.2 Active Networks Definition

Traditionally, the function of an intermediate node has been to forward packets from one end-point to another. Processing within the network was limited principally to routing, simple congestion control policy and uncomplicated quality of service (QoS) schemes. Their functions are rigidly built-in by node vendors that must follow designs dictated by slow standard committees (i.e., ten years or even longer) rather than taking rapid introduction of innovative, cost-effective, and user-demanded technologies. This type of a network can be regarded as "passive" network. Problems with "passive" network, however, have been identified [PSO99] and those are:

1. The difficulty of integrating new technologies and standards into the shared network infrastructure
2. Poor performance due to redundant operations at several protocol layers
3. The difficulty of accommodating new services in the existing architectural model.

Active networks exemplify how allocation visibility can be used for management purposes and also introducing new protocols and services without terminating system operations. Active networks (ANs) facilitate faster protocol innovation by making it easier to deploy new network protocols and services. These networks are "active" in two ways [TEN96]:

3. Routers and switches within the network is capable of performing computations on user data flowing through them
4. Users is allowed to 'program' the network, by supplying their own programs to perform these computations

3.3 Active Network Architectures

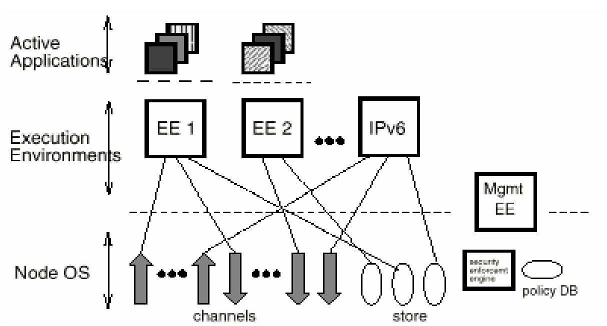


Figure 3.1 Active Network Architecture [CALV99]

The architectures are classified based on their approach toward the realization of active networking. Figure 3.1 shows the high level AN node architecture. When a packet arrives at a node, it is first handled by the

Node OS. Node OS verifies and identifies the content of packet. If the packet requires no further process it is then simply forwarded to the next node. If further process is needed, the packet is passed to the execution environment (EE). The further process in the EE may include running an active application or simply store the data brought by the packet.

3.3.1 Active Packet Approach

There are currently three architectures in implementing active network: active packets, active nodes and hybrid approaches. Most of the early active networks architectures follow "the active packet" approach. This scheme is fundamentally characterized by the fact that the code is carried inside the packet. The nodes are also active in a sense that they allow computation up to the application layer.

In active packet the code is carried by the packets. The code is either to be executed on the data of the same packet that carries the code or to be executed in order to change the state or the behaviour of the node. Examples of such architectures are the SmartPackets project proposed at BBN Technologies [SCH98], Active IP Option proposed at MIT [WET98a], and the M0 architecture proposed at the University of California at Berkeley and the University of Zurich [BAN98].

3.3.2 Active Node Method

In this approach, the packets do not carry the actual code or program. Instead, the packet carries some identifiers or references to functions reside in the active nodes. The packet, nevertheless, is considered active in a sense that they decide which functions are going to be executed on their data, and they provide the parameters for these functions.

Examples of active nodes architectures are CANES architecture proposed at Georgia Institute of Technology [BHA96a, BHA97], the DAN architecture proposed at Washington University and at ETH Zurich [DEC98], and the ANTS architecture proposed at MIT [WET98b].

3.3.3 Hybrid Model

Active packets can carry code efficiently only when the code is relatively simple and restricted. On the other hand, active nodes can efficiently provide any code given that the code is predefined. In the hybrid architecture, active packets carry actual code and other more complex code resides in active nodes.

A typical example is the SwitchWare architecture proposed at the University of Pennsylvania [GUN98], Netscript architecture proposed at Columbia University [YEM96].

3.4 Active Network Services and Applications

The most potential and important application of active networks stems directly from their ability to program the network: new protocols, and innovative cost-effective technologies can be easily deployed at intermediate nodes. Active networks are beneficial for a variety of specific applications. The following sub-sections present why and how is this technology possible for network management, congestion control, multicasting and caching applications.

3.4.1 Network Management

Active networks have potential to provide natural answer to the network management problem because management centres can be placed right in the "heart" of the network. This approach reduces both delays from responses and bandwidth utilization for management purposes. In addition, instead of waiting for a reply from a management centre, special code injected in the packets can act as "first aid" in case network encounters a problematic node. Packets can also be appointed to act as "guard" that constantly looking for anomalies as they traverse the network.

Three projects, among others, that apply active network technology concepts to network management are SmartPackets project [SCH98], the Network Management by Delegation Paradigm of Netscript Project of Columbia University [GOL98], and the Darwin Project of Carnegie Mellon University [CHA98a, CHA98b].

Active bridge [ALE98a, ALE98b] demonstrates how it is possible for management node to upgrade "old" protocol into "new" protocol on-the-fly. The prototype implementation on a Pentium-based HP Netserver LS running Linux with 100 Mbps Ethernet LANS achieves test tcp (ttcp) throughput of 16 Mbps between two PCs running Linux, compared with 76 Mbps unbridged. Test tcp (ttcp) is benchmarking tool for measuring TCP and UDP performance. Measured frame rates are in the neighbourhood of 1800 frames per second.

3.4.2 Congestion Control

Congestion control is a main target for active networking, since it is an intranetwork event and usually far away from the application. Congestion notification often takes a considerably long time to propagate from the point of congestion to the user host. As a result, there is a period of time during which congestion might have become worst – since applications have not learned or realized about it – or the notification arrives late that congestion and self-regulation is no longer needed.

By taking action at the congested node, active congestion control (ACC) scheme [FAB98] avoids this delay. The goals of the project are to implement congestion controls based on this idea and measure their performance improvement. Experimental simulations show that ACC congestion controls can improve performance by as much as 18%. Another example of applications is the Experimental Technologies Related to Congestion of the CANES Project of Georgia Institute of Technology [BHA96b].

3.4.3 Multicasting

The Internet and next generation of networks should be capable of handling a great variety of application traffics such as audio, video, and teleconferencing. Many of these traffics inherently require multicasting. New techniques have been continuously sought to provide the functionality of multicasting in an efficient, reliable, and scalable way. Active nodes can solve many current problems such as outlined in [PSOU99]: NACK implosion, concentrated load of retransmissions, useless transmissions, and duplication of packets.

Active Reliable Multicast (ARM) of MIT [LEH98] and Amnet Project of TU-Braunschweig [MET99] take advantage of active networking and suggest improvement schemes in the area of multicast techniques. The authors of CANEs [SAN01] develop NISTNet WAN emulators which allow delays, drops, and bandwidth limitations to be introduced on a link. These capabilities are necessary if the active protocols are to be useful. The topology contains two MPEG-2 video sources attached to active router via WAN emulators. Two receivers and are attached to active router. The active routers are connected to one another. The receivers contain streaming MPEG-2 hardware decoders and the active routers have dual processors. The video sources transmit MPEG-2 video in the half-D1 (360 x 480) format at full 30 frames per second. The data rate for a single video flow averages 2 Mbps with frequent bursts of up to 6 Mbps.

The active routers are reportedly capable of easily routing multiple multicast video streams from either source simultaneously to both clients while running the active applications. They demonstrate an obviously noticeable video quality difference between an active receiver that gets repairs from active routers, and a non-active one that only communicates with the source.

3.4.4 Caching

Traditional approaches for network caching place large cache at specific nodes in the network. The crucial point in these schemes is how to choose these specific appropriate nodes. For next generation networking technology, nodes should be smart enough to cache only objects that nearby clients will request in the future and to coordinate with each other to avoid caching the objects that already cached in neighbour nodes. The caching of objects at locations close to the clients can decrease both the network traffic and the time needed to retrieve the information. Active node may help to deploy mechanisms of coordinating the nodes.

A proposed scheme called self-organizing wide-area network caches [BHA98] of CANES Project of Columbia University is a sample research work in this field. The scheme considers the benefit of associating caches with switching nodes throughout the network rather than in a few hand-chosen locations. They propose the use of self-organizing or active cache management strategies for organizing active contents. They summarize their analytical and simulation experiments that the active mechanisms outperform all other methods (including Transit-Only caching) for correlated access. For non-correlated access Transit-Only scheme performs best in terms of average round-rip length. However, the AN schemes are always within 10% of the round-trip length of the Transit-Only caching scheme.

3.4.5 Safety and Security

Since Active Networks are much more flexible than passive one, a number of safety and security issues that need to be addressed are tremendously increased. Safety is mechanisms of reducing the risk of mistakes or intended behaviour, whereas security deals with the concept of protecting privacy, integrity, and availability in the face of malicious attack.

In active networks, active packets may misuse active nodes, network resources, and other active packets in various ways. In the same way, active nodes may also misuse active packets. Protecting the nodes and the packets in a flexible environment such as active networks is not an easy task. Some techniques [ALE98a, ALE98b, ARB98, SAN98] have been proposed to deal with these issues but the challenge is still wide opened.

3.4.6 Video-Quality Adjustment for Heterogeneous Multicast

Yamada et. al. [YAM02] proposed mechanisms for video multicast services in which diverse client requests are simultaneously satisfied while network resources are efficiently used. Their proposed mechanisms are developed based on the active networks approach. That is, intermediated nodes called active nodes adapt the video rate to the desired level. The mechanism is represented in Figure 3.2.

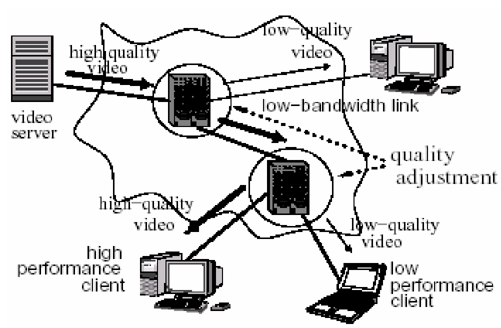


Figure 3.2 Heterogeneous Video Multicast

The proposed mechanism is implemented on processor-based programmable network equipment IXP1200. The IXP1200-based video-quality adjustment mechanism is evaluated in terms of the accuracy rate adaptation, the video quality, and the filtering performance. With support for multiple execution threads, the IXP1200 has an architecture suited for parallel data processing. High performance is achieved by distributing tasks among the StrongARM and microengines. StrongARM is general processor that can perform complex processing, whereas the microengines can execute a high volume of simple processing.

To find preferable distribution, they implement the filtering program on a PC and analyzed the process. For an 8 Mbps stream, they found that 75% the data is from the block-layer and that 56% of the processing time is spent in processing discrete cosine transform (DCT) coefficients. The code executed at the microengines cannot exceed 2048 programming steps and the number of registers is limited, but they are enough process block-layer data. All six microengines are used for video filtering so that evaluate the potential capability of the network processor-based video-quality adjustment can be evaluated.

The low-pass filter is intended for video streams in the MPEG-2 Program Stream format where video and audio streams are multiplexed into a single PS stream. An MPEG-2 PS stream used for evaluation is coded at the average of 8 Mbps. PS is product stream, such as DVD. A Video stream of 720x480 pixels and 30 fps, and audio stream of 192 Kbps are multiplexed into the PS stream. GoP length *N* is set to 15.

The highest layer of the hierarchical structure of MPEG-2 video data is called sequence layer. A sequence consists of several groups of pictures (GoPs). A GoP is a sequence of three types of pic-tures, I (Intra-coded), P (Predictive-coded), and B (Bidirectionally predictive-coded) pictures. A GoP starts with an I picture, followed by several P and B pictures. A picture is composed of 16-pixels height stripes, called slices. All sequence, GoP, picture, and slice layers begin with a 32-bit start code which is used for error recovery and for rewind and fast forward functions.

It is noted, however, that their implemented system cannot perform the video quality adjustment at the rate of 8 Mbps. The range of bandwidth their system can cover is from 8 Mbps to 2 Mbps. However, this is too high a requirement for client with poor bandwidth. In addition, another drawback is that the system performs adaptation at router level.

3.5 Application Level Active Networks

Many research efforts have been spent in the field of router level active network. However, given the numerous of security and management issues associated with router level active network it is highly doubtful if such approach will ever become reality [CAL00]. Network operators will be unlikely to permit third party programmes to run on their equipment without prior testing, and a strong guarantee that the programme will not cause security threat and degrade performance for other users. Since it will be extremely hard to create interesting programmes in a language which is important to enable termination guarantees, they regard this approach is unrealistic.

Application level active network (ALAN) [FRY99, PIA99] provides an environment for a dynamic, distributed execution and maintenance of networked applications at the application level instead of at router level. Thus the advantages of active networks are achieved, without the disadvantages of router level implementation. The main contribution of the approach is to enable rapid deployment of new communication services on demand by diminishing the weakness of current AN proposals.

ALAN proposes an improvement to the network performance by transcoding the streaming data format into a more compact one and hence saves bandwidth. Compressing the data as close to the client as possible and decompressing it at the destination are other options. In principle the codec and delivery system should be able to automatically adapt to the bandwidth available to the client without disturbing, for example, competing TCP traffic. ALAN concept [FRY99] can also be used to change the unsupported protocol to the supported one for applications heading to a network region which supports only that certain protocol.

Interesting and strong point for ALAN is that all the components involved can be accessed through their url addresses. This approach is not only simplifying the use of the system, but also makes ALAN system accessible through global Internet, which in turn supports its global availability.

3.6 Application Level Active Networks Architecture

ALAN System consists of regular client and servers such as WWW browsers and servers, located on the Internet or Intranet. Communication between servers and clients is enhanced by dynamic proxy servers (DPS) that are located at optimal points of the end-to-end path between the server and the client. Note that there may be more than one DPS involved in an end-to-end path.

In order to enhance the performance of an end-to-end communication, proxylets are downloaded into DPS. Proxylets are control modules which are downloaded onto the DPS. A proxylet may contain all the code required to performed enhancement process. These proxylets are obtained from Proxylet Servers. Another intention is that DPS can also download protocol stacks or protocol elements from Dynamic Protocol Servers. A significant benefit of downloading proxylets and/or protocol stacks from common servers is that code can be shared. The general overview of ALAN architecture is shown in Figure 3.3.

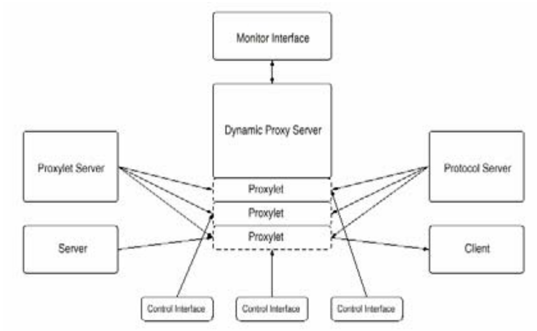


Figure 3.3 ALAN Architectural View

ALAN model allows adaptation to be distributed to the points in the network when it is required. It is the active codes in the Dynamic Proxy Server (DPS) that are shared by all the clients.

3.7 Application Level Active Networks Applications

The emergence of application level active networks (ALAN) encourages many researchers to explore the possibility of implementing the technology concept into current Internet architecture and obtaining the advantages sooner. The concept is also relatively free of problems of router-level active network, or known only as active networks, such as safety and complicated router and network management.

3.7.1 Application Layer Routing

An optimisation problem of ALAN is on how to appropriately place ALAN supported services in the global network. This is an application layer routing problem and requires a multi-metric observation such as available network bandwidth, topological proximity, user preferences and executive environment for proxylet (EEP) resources.

Authors in [GOS00] proposed an architecture that consists of four components: EEP discovery, routing exchanges, service creation and information routing. EEP discovery is capable of discovering the existence of all other EEPs in the global mesh. Routing exchanges create and retain EEP routing tables on the basis of various metrics. Service creation is the process whereby a proxylet or a set of proxylets are deployed and executed on one or more EEP. Information routing is the task carried out by proxylets and the information might include information transcoding, compression, TCP bridging or multicast splitting. They create an experimental web-based application named FunnelWeb that provides mechanisms to allow clients to choose an EEP or a set of EEPs on which to run proxylets based on one or more functions. However funnelWeb homepage cannot be accessed without any explanation. This is a discouraging fact to further consider this technique due to limited information available on its functionalities. They also do not provide explanation regarding the performance.

3.7.2 Policy-based Content Delivery

Authors in [LAR00] develop an architecture that allows individual user to define behaviour of application level active network by specifying content delivery or transcoding proxylet through user-defined policy. The approach makes use of WWW-based application level services that are associated with caches. The application addresses one drawback of initial ALAN purpose that is not coupled with caching infrastructure and hence does not gain the benefits of caching.

The contribution of the scheme is the provision of policy-based content delivery and caching mechanism that can be configured on either a system or user basis. The architecture of the approach is known as ConCEPT (Content Cache with External Proxylet Transcoding) cache. It makes five changes to the ALAN architecture. The modification includes the employment of user and system policies, policy manager, cache manager, use of support proxylets and intercache communication. However, no performance measurements are reported in the paper.

3.7.3 Video Conference

Authors in [BAL98] designs and implements an active network-based video-conferencing system. The system combines the paradigm of peer-to-peer and centralized video conferencing system. Their prototype is implemented using the Java language. The implementation of client relies on the tools *vic* and *vat* [CAN95] to generate respectively the video and audio streams. The stream playback is performed using the Java Media Framework (JMF) [SUNc].

On a conference a server that they call reflector customisability and scalability are addressed. That is, enabling the users to "upload" application code into reflector to change its behaviour and to customise it to their needs. Reflector can run on the intermediate node of the network and can use the information managed by the device to become aware of the status of the network and adapt to it. The system allows the reflector to migrate on different node as a consequence to adaptation. However, the concept and implementation is limited to simulate an active device by running mobile code on a workstation directly connected to a network device in a limited LAN environment. Unfortunately, they do not provide explanation regarding the performance of the video conference tool.

3.7.4 Distributed Adaptation for Complex Networks

Based on the concept of ALAN, authors in [YAR99] developed Conductor. Conductor is an integrated, adaptation, application-level framework that moves the responsibility for network complexity out of application and into the network. Unlike proxy solutions, conductor allows adaptations to occur at multiple locations in the network.

Conductor was developed on top of Linux 2.0. The framework is primarily written in Java. Adaptors are also written in Java. Conductor intercepts TCP streams generated by local applications through the use of loadable kernel module that allows a new set of functions for the TCP protocol Conductor makes use of the transparent proxy facility, present in Linux as part of the kernel's firewall feature, to discover the Conductor nodes between a client and server

All the results of the conductor's performance measurement are presented with a 90% confidence interval. They state that this network topology is a plausible and representative model for a mobile Internet access infrastructure. They use Pentium II 333 Mhz for network nodes and 2 Mbps waveLAN. 56 Kbps PPP serial connection is employed to emulate modem link and utilize 10 Mbps Ethernet between ISP and the destination Internet server. The experiments assume end-to-end response time, throughput and power consumption as the primary concerns.

Compression and scheduler adaptors are two functionalities for improving performance of this application. Compression adaptors can improve the end-to-end response time by reducing the size of image representations. The scheduler adaptor is responsible for turning on and off the network device on the mobile computer to save power while waiting for the arriving data stream.

Experiments consist of several configurations including the case without conductor, the case with conductor with no adaptor, and the case with conductor running with either or both scheduler and compression adaptors in place In the sample application scenario, the conductor implementation exhibits 30% increase in latency, 25% reduction in throughput, and 16% extra power consumption. The detailed measurement further indicates that using native threads, rather than user-level threads, will remove over 90% of the above overheads

Conductor dynamically deploys multiple adaptors to improve application's communication paths. Moreover, adaptor can be nested, deployed serially, or both. A planning algorithm is used in Conductor to decide which adaptor to deploy. However, it can be argued that the decision to shift the complexity out of application into the network is contrary to the principal concept of the Internet, which demands a simple work for network routers and switches.

3.8 AN and ALAN Comparison

To summary this chapter the comparison of active networks and application level active networks is presented as shown in
Table 3.1. In fact, most the mentioned projects of active networks are struggling to show the advantages offered at the implementation level. On the contrary application level networks have already shown their practicability. In addition, modification at the end stations and hosts are much more simple and direct than modifying intermediate router in the heart of the network.

To be applicable to the current Internet structure FANS follows the application level active network approach. This simplifies the problems of dealing with switch and router vendors and avoids lengthy process of setting up tunnels-like architecture and waiting for active network-supported switches and routers to be available.

Table 3.1 AN and ALAN Comparison

Active Networks	Advantages	Disadvantages
Router Level	• No modification to end station • Require less resources at end station and hosts	• Increase the complexity of the routers and their management • Security threats to the routers • Impracticable in the near future
Application Level	• No modification to the current Internet structure • Security threat to the routers is eliminated • Practicable in the near future	• Require more resources at end station and hosts • Require Modification to end station and hosts

Chapter Four

The System Architecture and Concept: Design Considerations and Implementation

4.1 Introduction

The current most popular and most widely used network protocol is Internet. The Internet, however, is not only well known for its simplicity but also for its unreliable nature of data delivery and, hence, lacking of QoS support. Interestingly, most of the research works, including mine, claim that their proposals are not to replace the Internet but to function complementarily with the Internet or even aimed at improving the performance of applications over the Internet. Therefore providing the satisfactory level of quality of service (QoS) over the Internet is one of the main focuses of FANS system and architecture.

FANS has been developed based on the experience gathered and studied from ALAN, service replication, application layer anycasting, application level Routing, grid computing, service location protocol (SLP), and the definition of active and QoS-aware middleware architectures for heterogeneous multimedia environments which are already described in Chapter Two and Chapter Three. Recall also that it is already pointed out in Chapter Two some concepts of FANS system architecture based on the recent development of networking technologies.

ALAN people have shown that their approach is worth for point-to-point communication through dedicated links between Australia and UK. However, it is not yet known any works that address a scenario in which many clients with their heterogenous bandwidth in countries spreading throughout global Internet joining ALAN-like system.

An intention of the improved active network approach is to improve the reliability of streaming multimedia applications over the heterogeneous nature of the Internet bandwidth. The performance improvement scheme is accomplished through the selective transcoding of the data flowing from media server to client. Furthermore, this work is intended to address also issue associated with an architecture that is appropriate for dynamic and scalable, QoS-aware and adaptable network computing and virtual organization.

The ability to trigger the activation of code resides in the middle node to perform the transcoding process when necessary is a key factor to the concept. To support the dynamic existence of peers and scalability of the system tracking mechanisms that are capable of maintaining dynamic list of active FANS members, including the one just leaving or joining the session, are developed. Peer machine consults the list when selecting an appropriate middle node in which the transcoding process occurs.

Servlet and Java class loader facilitate the traffic redirection and remote code activation. FANS transmitter and receiver are jointly managed by Java Media Frmework (JMF) and Java.net package. The FANS terminal are created using Java swing. Note that as there is no better way to explain the JMF as a significant component of FANS, some parts of this chapter regarding JMF (Sub-section 4.3.3 to 4.3.5) are cited directly from JMF documents [SUNc].

4.2 System Design Considerations and Goals

The improved system architecture explores the model of dynamic announcement of new services available in the network, dynamic replication of services, and dynamic server selection throughout the global Internet. The system architecture should also capable of handling the future virtual organization, of the grid computing paradigm, that may emerge between universities and research institutions. Anticipating those needs, the following design considerations and goals are taken into account.

4.2.1 Support for Streaming Multimedia Applications

As described in previous Chapters, an intended supported application is streaming multimedia applications between peer institutions. This kind of application stretches from teleteaching to video conferences. Further, this aspect can be extended to include e-University where various wireless devices with diverse capabilities exist in the University and may want to receive streaming applications from classes or shared media databases.

4.2.2 Support for Peers with Various Bandwidth Capacities

The peers factually have different bandwidth capacities. Therefore the developed system should consider this factor. That is, there will be no one media size fits all peer requirements. The system must be capable of delivering media in different size in accordance with the variety of bandwidth capacities of the peers. Peers are either dispersed throughout the global Internet or in Intranet environment such as e-University.

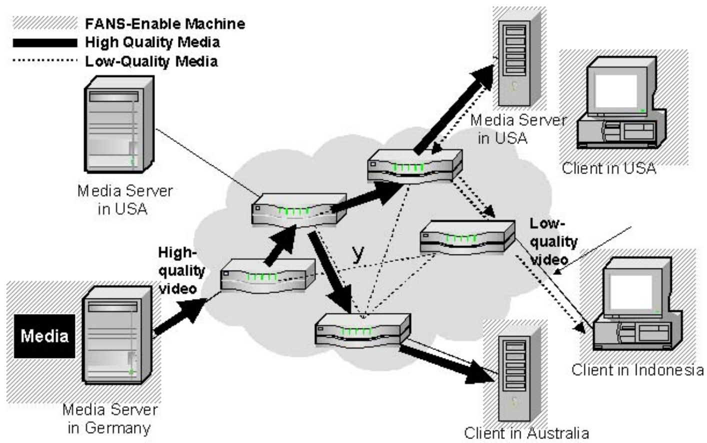


Figure 4.1 A Sample FANS Scenario

A sample of scenario is shown in Figure 4.1. When streaming a high quality video to clients with low capacity of bandwidth, the system proposes transcoding scheme to change the format of video to the more efficient one and if necessary to downscale the size of video presentation frame. If two peers in Indonesia and Australia retrieve a high quality multimedia education file from Germany, peer computer either in USA or Japan might operate as an active peer where the video quality adjustment can be performed. The selected intermediate peer performs the transcoding and downscaling process on the media for client in Indonesia. However FANS forward plainly and directly, the traffic heading to Australia whose sufficient bandwidth for original quality of media data.

The scenario shows how the system architecture is designed to promote QoS fairness. In accordance with their bandwidth capacities peer in Australia receive higher quality of service than peer in Indonesia. FANS should be able to adjust video presentation size and picture quality as the function of bandwidth capacity.

4.2.3 Selective Run-time Enhancement “in” the Network

Devices are connected to the Internet through a variety of access peripheral such fast ethernet switch and slow modem. The capacity of access devices differs in terms of bandwidth, which in turn creates heterogeneity of the ingress bandwidth to the clients. Therefore when creating multimedia streaming applications over the Internet, the data delivered should be customised to conform to the available access and client bandwidth. A process to customise the form of data to other format and size is termed as transcoding.

Transcoding process should be done at selective points, based on performance criteria, in the network as to optimally improve the performance. The demand of selective transcoding is supported by the fact that imposing additional processing is expensive for stringent delay. However, treating all the data pass through dynamic server equally is also inappropriate since the capacity of the network ahead a server is heterogeneous. Selective and consecutive transcoding processes, which are necessarily taken to adapt to the heterogeneity of peer’s bandwidth have not been supported and implemented in ALAN. FANS is intended to support this thought.

In addition, it is believed that the static nature of current ALAN system should improve by introducing resource monitoring and members tracking component to selectively perform traffic enhancement process on the best place. Resource monitoring decides whether an enhancement process is necessary. Members tracking system updates the member of FANS, which joins or leaves the session, and hence maintains the correct number of potential locations where the post-processing function can be done.

4.2.4 Support the Notion of Scalable, Dynamic and Mobile Peers

A weakness of original ALAN concept is that it is not scalable. The location of dynamic proxy server (DPS) has been a priori known and, hence, static. FANS has been developed as an effort to address, among other targets, this issue. FANS anticipates the growing number and dynamic nature of nodes participating in ALAN session. In this condition, users can join and leaving the session from anywhere and at any time. Therefore members tracking component is significant to maintain updated information regarding the current participating users in the FANS session. For this purpose, a requirement for scalable ALAN-like approach is that the participating users should agree to share their resources; be it cache, virtual machine or code. Only by sharing resources a fair benefit to all users can be guaranteed.

Addressing scalability issue means, among other definitions, to include technique dealing with dynamic announcement of new client joining the community and new services offered somewhere in the Internet. By realizing the existence of other peers with some resources to share, distributed works can be performed. One apparent advantage of ALAN, which is adopted by FANS is that all the participating clients, server and code can be referenced by its universal resource locator (url), or its domain name system (DNS) address.

FANS addresses the mobility and scalability issue by introducing a kind of directory service and service discovery component in its tracking component part. Directory service's job is to track newly connected users and services, and to maintain the list of active clients and codes. FANS employs service discovery method to locate and select the best service provider. This, in turn, supports also the mobility users. Users may move to anywhere and their location can be easily tracked and found as long as they have url or DNS address in their new place, and given they announce their new address through a registration mechanism.

4.2.5 Require No Changes in the Current Internet Structure

The unsuccessful tendencies of prominent approaches such as IntServ, DiffServ, and RSVP to address challenges in QoS-intensive applications are mostly due to their demand for additional structure or mechanism to the currently established and simple Internet infrastructure. At least, those technologies will have to wait for a long time for standardization process. Consequently, those techniques can not be applied in the near future and are very hard to be popular among network developers.

FANS is targeted to be available for implementation in the near future. Therefore the architecture must not demand any changes to the Internet infrastructure. In addition, the system must work in synergy with the Internet. These important philosophies influence the adoption of application level active network, the url-based reference to peers and media, and the selection of http-based tracking mechanism in FANS architecture.

4.3 FANS Supported Java-based Technology

Similar to other works in the field of active networks and application level active networks, FANS is built and developed using all-Java technology [SUNa]. Other reasons are due to the appealing features of Java, such as built-in multi-thread support, portability and cross platform compatibility. Java is, de facto, the ultimate language for Internet oriented application.

4.3.1 Java Class Loader

Class loaders are one of the cornerstones of the Java virtual machine (JVM) architecture. With class loaders JVM is capable of loading classes without knowing anything about the underlying file system semantics and, more importantly, allow applications to dynamically load Java classes as extension modules.

Class loader concept reveals the behaviour of converting a named class into the bits responsible for implementing that class. For the purposes mentioned in chapter two, FANS class loader complies with two requirements:

1. capable of loading classes from any legitimate url pointing to a class file
2. be able to load classes from any local, multiple class files since this is useful for testing or emulating an Internet server on a client, media server, and intermediate peer computer.

All Java virtual machines include one class loader, which is called the primordial class loader that is embedded in the virtual machine. The virtual machine assumes that the primordial class loader has access to a repository of trusted classes, which can be run by the virtual machine without verification.

4.3.2 Java Servlets

Java servlets [SUNb] are small, platform-independent Java programs compiled to a neutral byte code that can be loaded dynamically into and run by a host. Servlets are to the server what applets are to the client.

Deciding to use an extension mechanism for FANS, only two Java-enabled technologies are considered: Applet and Servlet. Unlike ALAN, which employs Applet together with remote method invocation (RMI) technology, FANS uses

Servlet. Servlet in FANS replaces Applet and RMI. Table 4.1 outlines the comparison of Servlet and Applet.

The choice is based on the following reasons:

1. FANS requires the capability to run code on the remote machine which are not supported by Applet. Applet can only bring the message from the client and display it on the remote machine: Servlets on the other hand reserve the permission to access database, changing file and even loading and run the file on the remote machine or server on behalf of the client.
2. Applet requires more significant amount of resource and time to upload into a machine than Servlets.
3. Servlets require no graphical user interface and therefore work as a background process.
4. Servlets capabilities support the notion of "programming the intermediate node on-the-fly" without requiring other support mechanism like RMI.
5. Servlets can forward requests to other servlets or services
6. Servlets can handle multiple requests concurrently and synchronizes the requests to support systems such as on-line conferencing

Table 4.1 Applets vs Servlets Technology

No.	**Applet**	**Servlet**
1.	Applet is a java program that runs in the appletviewer (a test utility for applets) or a WWW such as Netscape Communicator or Microsoft Internet Explorer	• Servlets are modules that extend request/ response-oriented servers • Servlets are the bodies of code that run inside servers. • Servers that can host servlets are Java-enabled servers that respond to client request
2.	Applets have GUI	Servlets have no GUI
3.	• Applet can not ordinarily read or write files on the computer that it's executing on • Applet can not make network connection except to the host that it came from • Applet can invoke the public methods of other applets on the same page	• Servlets are effective for developing web-based solutions that: • help provide secure access to a Web site, that interact with databases on behalf of a clien • dynamically generate costum HTML documents to be displayed by browsers • maintain unique session information for each client

4.3.3 Time-based Media and Java Media Framework

Time-based media is generally referred to any data that changes meaningfully with respect to time. Common forms of time-based media include MIDI sequences, movie clips, and animations, which can be obtained from a variety of sources, such as local or network files, cameras, microphones and live broadcast and multicast.

A crucial characteristic of time-based media is that it demands timely delivery and processing. Once a stream of media data begins, there are strict timing deadlines that must be satisfied, both in terms of receiving and presenting the data. Since it is delivered in a steady stream that must be received and processed within a particular timeframe to produce acceptable result, time-based media is most often referred to as streaming media. Figure 4.2 shows the media processing retained by JMF.

The Java Media Framework (JMF) [SUNc] is a Java Application Programming Interface (API) developed by Sun and other companies as an attempt to collide the portability and ease of Java with the appeal of multimedia. Unlike the Windows Media Player [MIC] and RealPlayer [REA], which have no custom graphical user interface, JMF is an excellent tool to create video conferencing or remote video education applications over intranet and Internet with custom user interface.

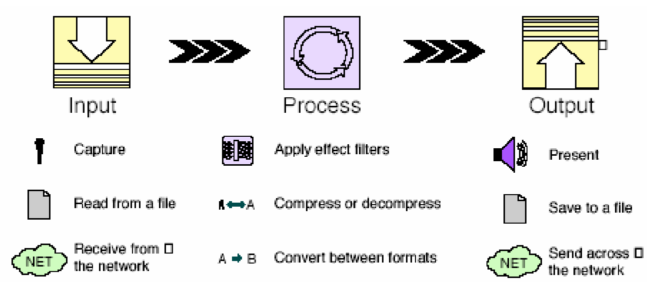


Figure 4.2 Media Processing Model [SUNC]

The term framework was used to the collection of APIs because its amalgamation of Java classes and interfaces, their inter-relationships, and extensibility considerations, all of which must be included in a group of APIs devoted to time-based media services. The objectives of JMF are to provide an APIs for time-based media that [IBM99]:

- is intuitive and easy to use
- is common across all Java platforms

- masks implementation techniques
- allows customisation of media processing
- manages resources such as microphones, speakers, cameras and display windows
- provides precise control over media capture, processing and presentation
- can synchronize multi-track or multi-channel playback
- satisfies application-specific performance and media quality requirements
- can be extended to support future technology advances and application requirements

4.3.4 High Level Architecture

Devices such as tape decks and VCRs provide a familiar model of recording, processing, and presenting time-based media. User provides the media stream to the VCR by inserting videotape. Then VCR then reads and translates the data on the tape and sends appropriate signals the television and speakers.

JMF employs the same model. The analogy is depicted in Figure 4.3. A *DataSource* is an abstraction of media protocol handlers. It encapsulates the media streams similar to videotape and a *player* provides processing and control mechanisms much like a VCR.

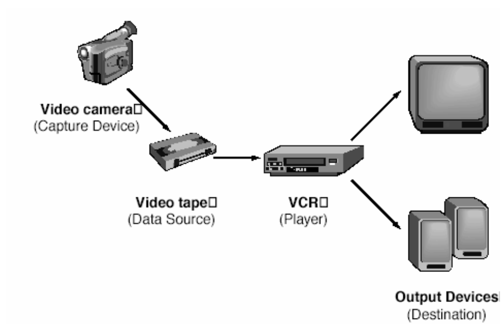


Figure 4.3 Recording, Processing, Presenting Time-based Media [SUNc]

DataSource and *player* are integral parts of JMF's high-level API for managing and controlling the capture, presentation, and processing of time-based media. JMF media players usually use DataSources to manage the transfer of media-content. A *Player* processes an input stream of media and renders it at a precise time. *DataSource* is used to deliver the input media-stream to the *Player*. JMF also supplies lower-level API that cares for the seamless integration of custom processing components and extensions. Figure 4.4 depicts the high-level JMF architecture.

Player is used to present time-based media such audio, video with JMF. To play multiple streams, separate *Player* is created for each one and *Player* objects is used to control the operation of others.

A *Processor* is a specialized type of *Player* that provides control over what processing is performed on the input media stream. Reformatting and transcoding the media file and resizing the media presentation size is accomplished by programmatically setting up the *Processor*. A *Processor* can output media data through a *DataSource,* so that it can be presented by another *Player* or *Processor*, and further manipulated by another *Processor*, or delivered to some other destination such as file or even to network.

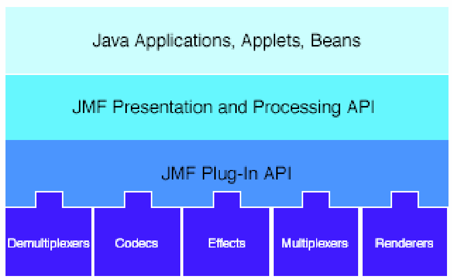


Figure 4.4 High-level JMF Architecture [SUNc]

The *MediaPlayer* is a Java bean that encapsulates a JMF player to provide an easy way to present media from an application. The *MediaPlayer* automatically forms a new Player when a different media stream is selected, which makes it easier to play a series of media clips or allow the user to select which media clip to play.

4.3.5 Extending JMF

To support the following capabilities:

- capturing and processing audio/video content
- customizing media players to produce audio/video special effects
- adding support for new codecs and media formats

JMF functions can be extended by implementing one of the plug-in interfaces to perform custom processing on a *Track*, or by constructing entirely new

DataSources and *MediaHandlers*. The available custom JMF plug-ins are *Demultiplexer, Multiplexer, Codec, Effect,* and *Renderer.*

Custom *DataSource* and *MediaHandlers* such as Players and Processors can be seamlessly integrated with JMF to support new format and incorporate existing media engines with JMF. *DataSource* can be saved to a file through *DataSink*, feed to player for presentation, or used to create another type of *DataSource* through *Processor*.

4.3.6 Real-Time Transmission Protocol (RTP) and JMF

Transmitting media data across the net in real-time requires high network throughput. It is easier to compensate for lost data than to compensate for large delays in receiving the data. This is very different from accessing static data such as a file, where the most important thing is that all of the data arrive at its destination. Consequently, the protocols used for static data do not work well for streaming data.

JMF supports RTP transmission and reception of audio and video streams. RTP provides an end-to-end network delivery services for the transmission of real-time data. RTP supports both unicast and multicast network sevice and, though it is often used over UDP, is network and transport protocol independent. Figure 4.5 shows the RTP architecture.

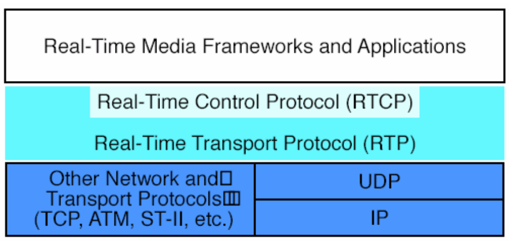


Figure 4.5 RTP Architecture [SUNc]

RTP provides means to identify the type of data being transmitted, determine what order the packets of data should be presented in, and synchronize media streams from different sources. Since RTP does not guarantee that data packets to arrive completely and in order, it is up to the receiver to reconstruct the sender's packet sequence and detect lost packets using the information provided in the packet header. However, RTP is augmented by a control protocol (RTCP) that provides control and identification mechanisms for RTP transmissions and, hence, to monitor the quality of the data distribution. Therefore if QoS is essential for a particular application, RTP can be used over a resource reservation protocol that provides connection-oriented services.

RTP applications are often classified into those that need to receive data from network (RTP clients) and those that need to transmit data across the network (RTP servers). Some applications such as conferencing and FANS intermediate peer do both, transmitting data at the same time it receives data from the network.

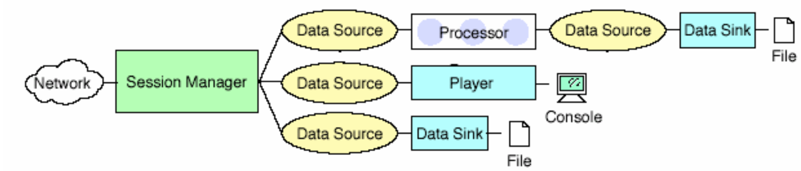


Figure 4.6 RTP Reception [SUNC]

Once obtained, the source cannot be reused to deliver other media. One can play RTP stream locally, save it to file or send it again to other node through the network. Figure 4.6 and Figure 4.7 show the reception and transmission schemes of the RTP. A *DataSource* encapsulates both the location of media and the protocol and software used to deliver the media.

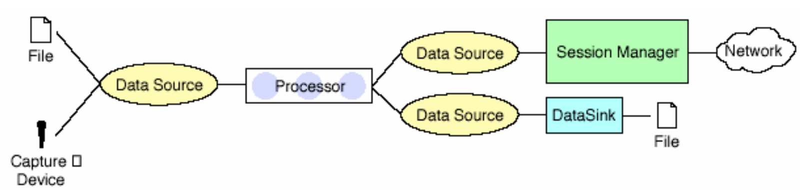


Figure 4.7 RTP Transmisson [SUNc]

4.4 FANS Architecture and ComponentsFANS follows the ALAN

approach to instantiate necessary codes on-the-fly and at the application level. The code could be activated at the best point in network, based on some performance criteria, to transform or customize multimedia data while streaming. However, because transforming and customizing code in real-time fashion is resource- and time- consuming, FANS introduces mechanisms to only selectively perform this process. For example, the transformation of big-sized code into a smaller one by compressing or transcoding the data format at an intermediate peer is useful and necessary only when the requesting terminal has limited, hence insufficient bandwidth to provide a continuous streaming of video applications. This process is, nonetheless, an unnecessary overhead cost if the requesting terminal owns sufficient bandwidth capacity to accept the original-sized data. To accomplish these functions the components that are described in the following sub-sections are developed.

FANS improves ALAN architecture by introducing member and service tracking mechanisms. Further, as ALAN architecture requires at least three elements namely client, dynamic proxy server (DPS) and web server to operate, FANS architecture unifies the function of client and server to peer machine. Moreover FANS diminishes the need of dedicated dynamic proxy servers since the activation of remote code is, in fact, performed at peer computer. This approach offers more efficient use of network resources such as servers for DPS. FANS approach can be considered as closer to the notion of virtual organization introduced by the grid computing philosophy, in which a group of institutions or personals agree to share resources for the common benefit. In FANS case, the resources are the memory (e.g., hashtable and vector cache) and Java virtual machine upon which the tracking mechanism and the remote execution of code rely. The system comparison of FANS and ALAN is shown in Table 4.2

Table 4.2 System Comparison: FANS vs ALAN

A L A N	F A N S
requires client,server, dynamic proxy server	a peer acts as client, server intermediate peer at once
intermediate node is known → static	intermediate peer is dynamically updated
is unscalable and has no mobility support	supports the notion of scalable and mobile system
enabling technology: Applet, RMI	Enabling technology: Servlet, Jswing

Figure 4.8 shows FANS general architecture. On the left side of the figure is the tracking architecture. In general the architecture of FANS is divided into two modules: media handling component and tracking component. Media handling components deal with the transferring, processing, reformatting, rescaling and presentation of audio/video media. Members and media tracking elements are responsible to dynamically maintain two lists, namely member tracking list (MmTL) and media tracking list (MeTL). The active environment is the heart of the system, and works as background process. It unifies all of other system components. The active environment is developed using Java virtual machine and Servlet server.

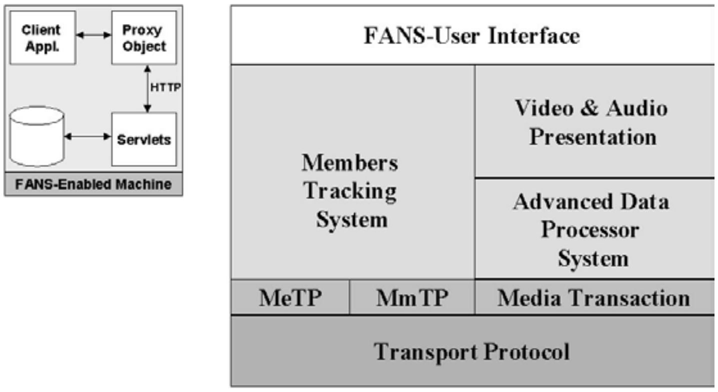


Figure 4.8 FANS Architecture

Procedure of determining the necessity to add media post-processing at intermediate peer and of selecting an appropriate intermedia peer to perform the processing is carried out by the following routines:

1. Gathering information regarding the bandwidth capacity of each client or peer
2. Collecting reports about the location and the size of multimedia files that are available on the session
3. Identifying the need to include intermediate post processing in the process of streaming applications from media source to requesting client. This is done by calculating and comparing the bandwidth capacity of requesting node and the media file size
4. In case of an intermediate media post processing is necessary, the system consults the list of actives peers to select one with the highest capacity or with the capacity higher than media transmission requirement. This is to ensure that the media is delivered in the best quality possible.

4.4.1 Media-Handling Component

The media-handling component consists of transport protocol, media transaction, advanced data processor system, and FANS terminal for audio/video presentation. The FANS terminal is provided for interfacing the system with users.

The transport protocol component is responsible for handling RTP communications between peer computers. Media transactions take over the media from transport protocol and decide whether the post processing of the media is necessary. When required, the transaction component forwards the media to the advance processor for further reformatting, rescaling and transcoding process. The audio/video presentation and user interface components are the top level of the system whose tasks are to accepting media from the lower level and taking the input from user’s actions.

The processing, reformatting, rescaling and presentation of media are implemented by means of JMF. Servlet and Java class loader facilitate the traffic redirection and code activation at the remote machine. FANS transmitter and receiver are jointly managed by JMF and Java.net package. FANS uses RTP over UDP as the protocol for delivering data and utilizes networking API provided by JMF for this protocol. FANS terminal is created using Java swing.

4.4.2 Tracking Components

Tracking components and procedures are implemented using Servlet and Java hashtable. Member and media tracking elements are responsible for media and member discovery and updates. These particular parts work dynamically to deal with new member and media registration and to correctly update the two lists, namely the member tracking list (MmTL) and media tracking lists (MeTL). The dynamic nature of the lists is caused by the dynamic event of members leaving and joining FANS session.

The member’s tracking system works as follows. Each peer computer maintains two lists. Members tracking list contains information about the [ip:port] pair and bandwidth capacity of the currently active peer computers or FANS members. In addition media tracking list stores the characteristics of multimedia file, such as the media size and location, that are available on the peer computers currently listed on the members tracking list.

List update occurs when peer computers join and leave the FANS session. Once a peer computer leaves or joins the session the list of the available multimedia media is updated as well. Therefore there is interdependency between MmTL and MeTL.

Figure 4.9 shows the architecture of tracking protocol and mechanism. HTTP protocol is used for exchanging data for tracking and updating purpose. The existence of a peer computer and multimedia file is notified by its url address i.e, [ip:port] pair. Service registration point (SRP) is a list located in a main peer computer. SRP is acting as central list. A peer computer announce its assistance by sending a packet signal containing its [ip:port] pair, bandwidth capacity and country origin. In return, SRP sends all the available, currently active or on-line FANS members and their shared media file. Each peer computer periodically sends beacon signal to SRP to inform that it is still active or on-line and to obtain any updates.

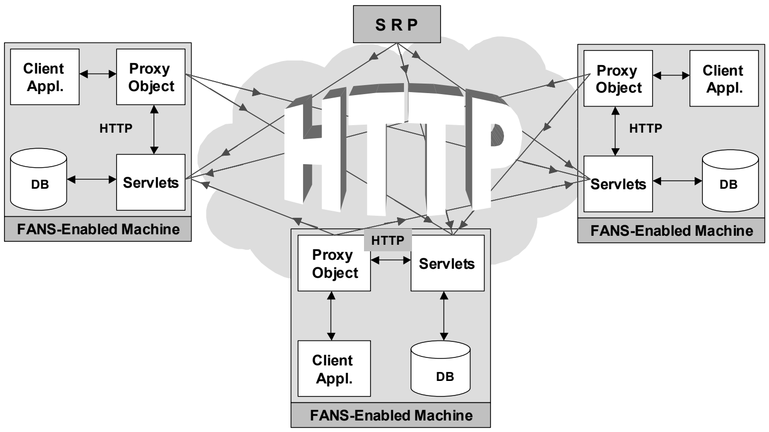


Figure 4.9 FANS Tracking Protocol and Mechanism

In short, the functionalities of FANS tracking system are as follows:

1. New member and media registration
2. Information update in case of member leaving and joining the session
3. Monitoring the existence of all members and media
4. Consultation function is terms of the decision on the involvement of intermediate peer and its selection
5. Key to the potential scalability factor of FANS

4.5 FANS User Interface

The improved concept and architecture incorporate also a user interface for dynamic multiple locations streaming multimedia environment. While FANS is intended to be capable of reaching the client with poor bandwidth, it is also imperatively important to design a user interface of the system that is simple, self-descriptive and encouraging for the users. The usability of a multimedia system like FANS depends significantly on the design of the user interface. That is, the audio-visual display and required actions or procedures that manage user interaction with the system. Computer system with poorly designed user interface discourages users from making use of it.

Two major aspects drive the FANS design process. Those are the working mechanisms of FANS, and the intended audience and environment that are potentially utilized the system. FANS has been developed with mechanisms to support distributed multimedia system where peer computers share resources for their common benefit. The intended users are those who at least familiar with the Internet browsing procedure. The potential applications of FANS include, but not limited to, distance education, pervasive computing, mobile device and ad-hoc environments.

It is identified that to increase usability of a system that supports dynamic multiple location users that agreeably share their resources, as is the case of FANS, the following attributes should be taken into account:

1. Self-descriptive user interface
   All components of the user interface is designed and given text attributes, which are self-explanatory.
2. Preserving setting and attributes across computers and their environments
   Each peer computer possesses its own setting and attributes. However to work in synergy and cooperatively the attributes are announced to other peer clients.
3. Synchronizing the dynamic data automatically and at run-time
   Peers are free to join and leave the multimedia session at any time. FANS maintains two lists which are updated dynamically based on leave and join processes. Two components of the user interfaces monitor and display the update information.
4. Hiding complexity from technically incompetent users
   Users pay attention to media and control part of the user interface. To be able to operate the control part of the user interface a familiarity with ordinary Internet browsing is sufficient.
5. Providing access and information to the more technical data for more knowledgeable users
   A DOS-based window component is available for users who are interested to track and monitor the treatment made to the data flowing through media source, via intermediate peer and return to the requesting peer.

The user interface is divided into control part and media part. The media part deals with audio-video presentation and simultaneously hides the complexity of media stream handling such as media acquisition, transmission and synchronization. The control part handles the originality of an application. It controls the media part and interacts with users.

The user interface matches the self-descriptive principle by putting clear, self-explanatory text attributes and providing fill-in form text area when necessary. Two panel lists are available to allow users to select the multimedia file name and media source address.

4.5.1 The Media Part

The media part is where the system presents the video and audio. In FANS, the size of the media part is constant but the quality and frame size of the video presentation vary.

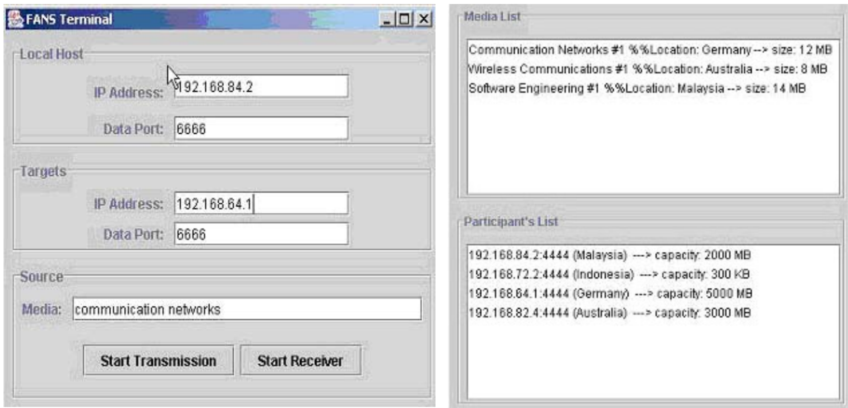
FANS distinguishes the members based on their bandwidth capacity. The more bandwidth a client has the higher the quality it can receive and the bigger the frame it can show, and vice versa. Figure 4.10 compares the presentation of full frame and half frame of a video application.

Figure 4.10 Full-scale and Half-scale Movie Presentation

4.5.2 The Control Part

This is the part of the user interface where the users specify the media they want to retrieve and to initiate the media delivery. The control part gives user updates and information of all active peer computers and their media. The control part is divided into static and dynamic area and is shown in Figure 4.11.

The static area consists of the information about the automatically detected host computer the users use, the source address from where they want to retrieve the media and the name of the media. In case a user has very limited idea about the source and the media, they can select one from the lists available at the dynamic area of the user interface.



(a) (b)

Figure 4.11 Control Components: (a) static part, (b) dynamic part

The dynamic area has two functions that conform the design principle: preserving setting and attributes across computer and their environments, and synchronizing the dynamic data automatically and at run-time. The media info contains information about the media filename, size of the media and the ip address and the country where the media is available. The participant's list records the currently active peer computers and their bandwidth capacity. The information that is displayed at the media info and participant's list is dynamic and updated accordingly when a peer leaves or joins the session.

Figure 4.12 FANS User Interface

Figure 4.12 shows the complete FANS user interface. It is an integrated unit of static control part, dynamic control part, and media display part.

FANS consistently hides the complexity of traffic redirection, video frame rescaling and media transcoding processes from most of the users. However, users with more technical background can track the process through the DOS-based log terminal that run as background process. These policies satisfy the FANS design consideration to "hide complexity from technically incompetent users", but at the same time "providing more access and information for the skilled users".

4.6 Comparing FANS Architecture and Other Middlewares

Table 4.3 outlines the comparison of the concept and architecture of FANS with active middleware and QoS-aware architectures described in Chapter two. The middleware components can be classified into seven important functionalities for a system whose function is to support QoS-aware environment. FANS architecture offers the most complete functionalities to deal with Qos intensive applications such as streaming multimedia.

Table 4.3 Comparison of FANS Architecture and Other Middleware Concepts

Components	Qos-Aware Middleware	Active-middleware	FANS Architecture
Presentation/GUI	X	√	√
QoS/Resource Manager	√	√	√
QoS/Resource Adaptor	√	√	√
QoS/Resource Monitor	√	√	√
Tracking Algorithm	X	X	√
Active Environment	X	√	√
Protocol Handler	√	√	√

FANS graphical user interface is described and shown in Section 4.5. FANS tracking components explained in Sub-section 4.4.2 covers the tracking algorithm and QoS monitoring function of the middleware concepts. Protocol handler in FANS is part of media handling components and its functionality is demonstrated in Sub-section 4.4.1. The functionalities of QoS/Resource Manager are handled by the active environment and are described throughout Section 4.4. Active environment is accomplished as a combination work of Java virtual machine and JMF.

Chapter Five

Defining QoS Metrics and Setting Up the Experimentation Testbed

5.1 Introduction

Transmitting interactive video over packet switching networks like the Internet is time-constrained. If a packet containing a video frame is not delivered before its scheduled play-out time, and the frame cannot be displayed accordingly, the overall quality of video degrades and the information might lost. The difficulty of meeting this constraint lies mainly in frequent occurrences of packet loss, which are commonly caused by network congestion that might occur due to the bottleneck access point. The recovery of lost packets before the display times is not always practical because of the delay associated with detecting and repairing the lost packets.

When one tries to measure streaming audio and video quality on the Internet, a flood of factors occurs. Factors like late and lost packets, rebuffering during playback, bandwidth of transmitter and receiver add to the complex mix of measurements to evaluate the overall streaming audio and video quality. Furthermore, streaming media content over the Internet itself is complicated due to the following aspects:

1. Multiple data types are involved (audio & video)
2. The data is much more sensitive to latency
3. Different default delivery protocols are involved (User Datagram Protocol (UDP) vs. TCP/IP)
4. Multiple technologies are involved (Real, Windows Media, QuickTime).

To evaluate the reliability of the new concept and architecture a LAN-based WAN emulator has been developed. The WAN Emulation for FANS incorporates the use of three end nodes representing three different channel qualities for each connection. The three nodes are noted as the source peer, the destination peer and the intermediate peer respectively. A router is assigned as WAN Emulator running FreeBSD [BSD] with DummyNet [RIZ97] feature. The DummyNet introduces processes called pipe and queue to represent a channel with defined bandwidth, transmission delay, queue size, and packet loss rate.

5.2 Identifying QoS Metrics for Streaming Applications

Because of the large performance variations that occur on the Internet, it is important to measure the performance of the system to gain an objective view on how the media performs at the requesting peer. Three major factors that affect streaming quality, which need to be measured, are startup time, audio quality and video quality. Within these factors are specific elements, such as connect time, redirect time, initial buffer time, frame rate, recovered, lost and dropped packets, and bandwidth variation.

Startup time is the time it takes when user presses, for example, the play button until the clip begins. Startup Time is the sum of time required for initial connection (including DNS searching & time to first byte), redirection Time, and Initial Buffering.

a. Initial connection is the time taken to establish a real time streaming protocol (RTSP) connection between the streaming server and the streaming peer.
b. Redirection is the time needed to transfer data to the last server from a second and subsequent servers, to which the first server may redirect data
c. Initial buffering is the time required to start viewing and hearing a streaming media clip from the time the data arrives in the buffer of the peer computer

Audio quality is derived from audio encoding and audio delivery mechanisms of the system. Audio encoding can be divided into the number of audio channels, bit rate per channel and quality of original content. Audio delivery can be divided into the delivered bandwidth and packet delivery.

Video quality can be obtained from video encoding and video delivery. The higher the successful delivery and the encoded bit rate for both audio and video, the higher the quality. Video encoding can be measured as the encoded bit rate, encoded frame rate and the quality of original content. Video delivery comprises the delivered bandwidth and packet delivery.

Two more factors that contribute to the quality puzzle are encoding quality and delivery quality. Encoding quality is based on a complex set of decisions and tradeoffs depending on the potential audience: how much to dedicate to video (including frame rate and window presentation size) and how much to dedicate to audio (one channel mono, stereo, surround sound), video resolution and video frame rate. The tradeoffs become even more difficult when the audience is going to be a mixed one in terms of bandwidth capacities.

Delivery quality includes bandwidth, packet and frame rate data:

1. Average frames per second (video only) that indicates the average number of video frames received. This can be compared to the average number of frames actually streamed. Television and movies display video at 30 frames/sec, which is the rate at which humans discern full motion.
2. Late and lost packets factors. Late packets, received by the client's buffer but are too late to use, is the worst case scenario. Not only can the client not use the packets, the packets use up bandwidth on the receiving side and taking up space that could be used for other usable packets. Lost packets are never a good effect to the client. Both late and lost packets have very undesirable effects on audio and video. The effects include jitters, frozen video, and audio static. By improving connectivity to areas that are experiencing bad connection, the number of late and lost packets can be reduced.
3. Average audio and video bandwidth factors. The negotiation that occurs between the server and the computers at the receiving end to minimize packet loss. With connection problems or insufficient bandwidth, the server will scale back, or adjust the stream. The server would rather scale back and deliver fewer frames than drop frames. Adjustment can appear as a smaller viewing window as well as slower video motion.

The following elements are four most common delivery phases that affect the quality of a stream:

1. During the initial negotiation between the server and client, if there is not enough network bandwidth or less than optimal network conditions, the streaming server will decide to not deliver the ideal number of packets and will scale the stream back or to scale down the window presentation size to stream fewer packets.
2. After playback begins, if network conditions degrade the server will again make a decision to scale back the number of packets or to scale down the window presentation size it attempts to deliver.

3. Packets arrive to the client too late for the software application to use them
4. Packets get dropped or lost on the way and never reach the client computer.

5.3 Adjusting the Media Quality to Cope with Bandwidth Condition

The quality of service for streaming media applications over the Internet depends on the size of media file being streamed and the bandwidth capacity of the network. There are some key factors that determine whether or not media content will fit within a certain amount of bandwidth:

- Frame size. The bigger the image sizes in pixels, the more bandwidth it will take for the content to play. In order to keep the picture size large, some of the following factors should be to taken into account.
- Frame rate. The higher the frame rate, the more bandwidth it will take to play back the content. In order to keep the frame rate high, it is important to regulate frame size, picture quality, and/or number of colours.
- Picture quality. The higher the picture quality, the more bandwidth it will take to play back the content. In order to keep the picture quality high, one needs to adjust frame size, frame rate, and/or number of colours.
- Number of colours. Sometimes limiting the palette of colours in creating content will help conserve bandwidth. To maintain bright, rich colours, though, the above factors should be controlled to fit the content over low bandwidth.

At very poor bandwidths, the considered actions include sacrificing the video altogether and offering audio-only feeds. At higher bandwidths, of course, much more freedom to include richer audio and video, and even to add script events and other data to make the feed still richer are possible.

FANS is concerned with the “perceptual” quality of the actual audio and visual content on delivery. The elements that determine the perceptual quality of the video can be categorized into two aspects, i.e., the motion of picture and the quality of picture presented. It is believed the motion of picture plays more important effect to the perceived user satisfaction than the quality of picture presented. The second aspect is acceptable, at least, with the fair quality. However, rigid or broken motion of picture causes the meaning of the video is partially lost or the sent message is potentially non-understandable.

5.3.1 Relation of Bandwidth and the Quality of Motion Picture

The video update rate, or frame rate, is measured in "frames per seconds" (fps) and corresponds to the number of quick succession still pictures shown in sequence every second to create the illusion of a moving picture. The more frames per second (fps), the smoother the motion appears. Television in the U.S., for example, is based on the NTSC format, which uses 30 interlace frames per second (60 *fields per second*). British televisions have a frame rate of 25 fps. In general, the minimum fps needed to avoid jerky motion is 30 fps. Some computer video formats, such as AVI, provide only 15 frames per second.

It would seem obvious that higher frame rates lead to a better quality moving picture. However with the very limited bandwidth, this is not always the case. The problem is that there is only fixed bandwidth, or amount of data, that one can use each second. Therefore if the number of frames shown in every second is doubled, the amount of data one can use for sending each frame is halved. In other words, one gets more pictures per second and a smoother movie, but the quality of each still picture will be reduced.

In an Ethernet environment the maximum video update rate can be calculated as follow:

The max # fps = Ethernet Data Rate(bps)/Total Frame Physical size (bits) (1)

e.g. Max. # fps = 10.000.000/ (84x8) = 14.880 fps
where (84x8) is the smallest size of Ethernet size

The maximum throughput is obtained with the largest frame (1526 Bytes) plus 12 Bytes equal to inter-frame gap. So the total utilized period is 1538 Bytes. Further, with the formula (1), the frame rate is 812,74 fps.

The link layer throughput = Frame Rate x Size of Frame Payload (bits) **(2)**
= 812,74 x (1500x8) = 9.752.880 bps

If the maximum throughput of Ethernet is 10.000.000 bps then with the maximum frame size the throughput efficiency is 97,5%.

At poor bandwidth environment, reducing the frame rate is an option to save bandwidth. Bandwidth requirement to deliver a motion picture can be calculated with the following formula.

$$BW = \#\ \text{Bits per pixel (bits)} \times \text{Dimension} \times \text{Frame rate} \qquad (3)$$

Number of bits per pixel determines the quality of colour picture. For example, grey-scale picture requires 8 (eight) bits per pixel. The higher, colour quality might require 24 (twenty-four) bits per pixel. With the standard dimension of 320x240, the bandwidth requirement for delivering video with 30 fps is:

$BW = 8 \times (320 \times 240) \times 30\ fps = 18{,}432\ Mbps$ - grey-scale picture

$BW = 24 \times (320 \times 240) \times 30\ fps = 55{,}296\ Mbps$ - colour picture

It is shown that up to 66,7 percent of the bandwidth consumption can be saved by reducing the colour quality of a video delivery. When choosing to reduce the frame rate instead of the colour quality of the picture then the following efficiencies are exposed:

$BW = 24 \times (320 \times 240) \times 30\ fps = 55{,}296\ Mbps$ - smooth motion

$BW = 24 \times (320 \times 240) \times 20\ fps = 36{,}864\ Mbps$ - satisfactory motion

The above sample gives 33,3 percent of bandwidth saving.

5.3.2 Relation of Bandwidth and the Quality of Picture Presentation

In uncompressed form, video sequences require a massive amount of bit rate. In principle, for every frame pixel, 3 bytes have to be reserved for its RGB colour values. However, as the human eye is known to be more sensitive to luminance rather chrominance, a sub-sampling technique may be used when using the YC_bC_r colour space [JAN02], with Y standing for luminance and C_b and C_r for chrominance difference value. In the latter colour space, it suffices to store only a C_b and C_r a value for every 2 pixels and is often referred to as 4:2:2 sub-sampling. For Standard Intermediate Format (SIF) sequences of 352 x 288 pixels and 25 frames/s, this leads to a bit rate of 352 x 288 x 25 x 16 bit/s = 40.55 Mbit/s. To reduce this bit rate, video compression techniques can be used.

Authors in [KOR00] stated that even with the perceivable degradation of media quality user's might be satisfied in some applications, especially if his/her need for data size reduction is significantly higher than his/her need for original source quality. This is very often the case in real applications. It is acceptable to use the maximum compression as long as the message can be brought through.

Streaming video services require the effective bit rate of the video stream (i.e. the bit rate of the encoded video + some packetization overhead) to be such that it fits within the access capacity of the user. Otherwise, information will be lost and the expected video quality will not be reached.

5.3.3 Quality in terms of the Size of Video Presentation

A larger image uses more bandwidth per view. By reducing the frame presentation size of video the media can be delivered in a more efficient manner. Referring to formula (3), delivering grey scale video with 320x240 frame size requires:

BW = 8 x (320x240) x 30 fps = 18,432 Mbps

However, if the video presentation frame is downscaled into half of the original then the bandwidth requirement is:

BW = 8 x (180x120) x 30 fps = 5,184 Mbps

This requirement equals to only 25% of the requirement with original frame presentation size.

5.3.4 The Effect of Packet Lost and Delay

The effect of packet loss is far-reaching. Packet loss degrades not only the quality of the frames contained in lost packets, but also the quality of their subsequent frames. This is because of motion compensation employed by video codecs. Motion-compensated video codecs remove temporal redundancy in a video stream by encoding only pixel value difference (prediction error) between an image to be encoded (inter-frame) and a previously transmitted image (reference frame). Packet loss can introduce an error in a reference frame, which can be propagated to its subsequent frames and get amplified as more packets are lost.

Error propagation can be controlled by more frequently adding intra frames (which are coded temporally independently). However, the ratio of the compression efficiency of an intra-frame over an inter-frame is as large as 3 to 6 times. Increasing the frequency of intra-frames could increase the bandwidth requirement too much for video transmission over a bandwidth-constraint network. Nonetheless, the severe degradation of image quality due to error propagation has forced several popular video conferencing tools, such as vic [KOU97] and CU-SeeMe [CUSM], to adopt an even more drastic approach. Using a technique called conditional replenishment, these tools filter out the blocks that have not changed much from the previous frame and intra-code the remaining blocks. Since all the coded blocks are temporally independent, packet loss affects only those frames contained in lost packets. However, this enhanced error resilience comes at the cost of low compression efficiency. Additional compression can always be obtained if temporal redundancy is removed from each coded block (i.e., by coding only their prediction errors).

5.4 Out-Service and In-Service Metrics

Naturally, the benchmark for any kind of video quality assessment is subjective experiments, where a number of people are asked to watch test clips and to rate their quality. The problem with subjective experiments is that they are time-consuming, and hence, expensive and often impractical. Another simple error measures such as Peak Signal-to-Noise Ration (PSNR) operate solely on the basis of pixel-wise differences and neglect the impact of video content and viewing conditions on the actual visibility of artefacts at the user side. Therefore, their predictions often do not agree well with perceived quality of the users.

Another way to perform objective measurement of data transmission is measuring packet loss ratio (PLR), bit error rate (BER), and other network-related parameters such as latency. Establishing and maintaining a certain level of network quality of service (QoS) for different applications is a very active and challenging research area at the moment. However, the measurements and protocols used there are unacquainted to the actual content being transmitted over the network as well as have no direct relation to the video quality as perceived by the user.

Two different approaches of the more advanced perceptual quality metrics can be distinguished as follows. First are the human visual based approaches. These methods are based on models of the human visual system, and are the most general and potentially the most accurate ones. However, the human visual system is extremely complex, and many of its properties are not well understood even today. In addition, implementing these models is computationally expensive due to their complexity. The second approaches use metrics that are not necessarily relaying on general models of the human visual system. They can exploit a priori knowledge about the compression and transmission methods as well as the pertinent types of artefacts using ad-hoc techniques or simple specialized vision models. While such metrics are not as versatile, they normally perform well in a given application area. Their main advantage lies in the fact that they often permit a computationally more efficient implementation

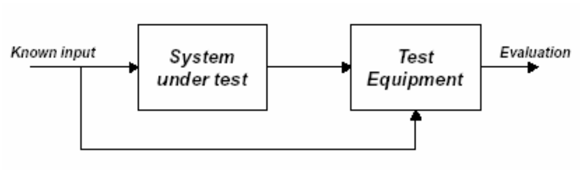


Figure 5.1 Out-of Service Testing for a System [WIN01]

The emphasis of most metrics today is out-of-service [WIN01] testing. This approach demands the full reference video to be available to the metrics. This is quite a severe restriction on the kind of applications such as a metric that can be used for, however. The schematic system for this testing is shown in Figure 5.1.

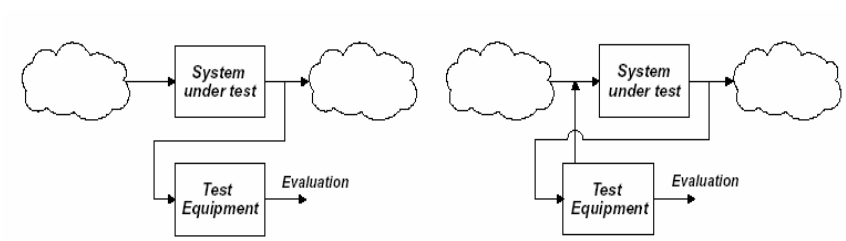


Figure 5.2 Non-Intrusive (l.) & Intrusive (r.)
In-Service Testing Setup [WIN01]

In-service metrics [WIN01] are designed to monitor and control systems while in operation. This approach is much more powerful and, hence, is used in FANS quality measurement. They can be used to carry out measurements at a practically any point of the transmission chain. This is a particularly important issue in multimedia streaming applications. The setup can be intrusive or not, depending on the objective of the test and the nature of the testing methodology. This approach is depicted in Figure 5.2.

The corresponding metrics are often referred to as reduced reference and no-reference metrics, respectively. Most of the proposed methods are aimed at identifying certain features in a scene and assessing their distortion. They also take into account knowledge of the compression method and the corresponding artefacts [YUE98].

5.5 WAN Emulator for System Experimentation

The desired WAN Emulation for FANS incorporates the use of three end nodes connected by three different channel qualities among each other. These three nodes are noted as the source node, the destination node and the intermediate node. For simplicity, each node is named consecutively after Germany, Indonesia and Malaysia. Figure 5.3 shows the description of the WAN connection.

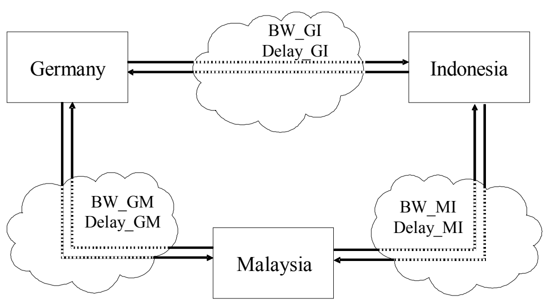


Figure 5.3 Desired FANS' WAN Model

5.5.1 Construction of WAN Emulation Model

The construction of WAN laboratory size model is performed by using an emulation system with three of its network interfaces connected to each end node. Figure 5.4 shows the physical setup of the emulator to construct the desired FANS's WAN model. The WAN Emulator, which is represented with a WAN cloud is intended to be constructed using some packet processing devices that are able to emulate as close as possible, the real network behavior on the Internet. Considering the limitation of resources and the concept of simplicity, it would be preferable to have an emulator that comprises of a single machine with great flexibility for its configuration.

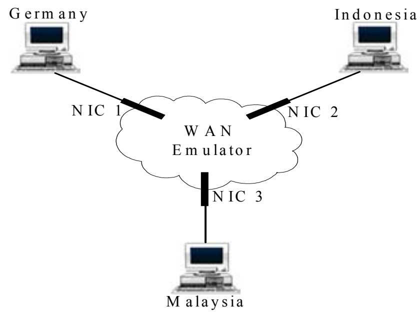


Figure 5.4 Physical Setup of FANS' WAN Emulation

A FreeBSD-based emulator is created, by running dummynet functions in its kernel. The system is also configured to run as a router with IP forwarding capability, thus operates at IP layer.

5.5.2 DummyNet

The original design of DummyNet [RIZ97] allows sophisticated network emulation and network protocol/application testing performed using only a single machine. The simple yet very effective approach that utilizes the multi-layered characteristics of computer network architecture enables this concept. The DummyNet introduces processes called pipe and queue to represent a channel with defined bandwidth, transmission delay, queue size, and packet loss rate. These configurations are keyed into the operating system using ipfw command. For detailed instruction on how to use the ipfw command, please see "man ipfw". In order to pass the packet into the pipe, packet filtering performs a mechanism to provide desired treatment to every incoming and outgoing packet. Figure 5.5 shows the illustration of where the packet filtering occurs in the packet processing within the Free-BSD box.

Every time a packet is transferred between link layer to the upper layer or between interfaces, it passes through the filter rules, which then checks the properties of the packets over which rule that matches, and performs operation on the packet itself. The way filters observe and process the packets depends on the *sysctl* variable (*net.inet.ip.fw.one_pass*) in the Free-BSD system. When the *net.inet.ip.fw.one_pass* is set to 1, the packet will pass through the first matched filter only and then continues to the next layer or interface. If the variable is set to 0, then the next filter after the first matched will be observed and so on. This of course introduces two different mechanism of packet filtering to occur in the system. These mechanisms are show in Figure 5.6.

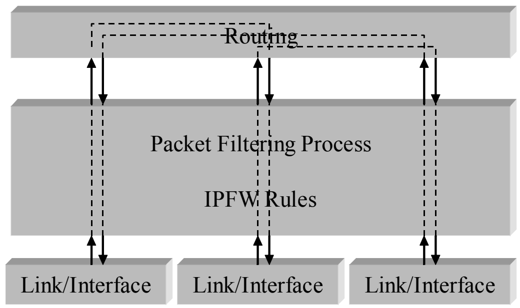


Figure 5.5 Packet Filtering layer used by Dummynet

5.5.3 System's Preparation

In order to enable DummyNet on the Free-BSD machine, after the installation of FreeBSD on an Intel Pentium II-250MHz, 128MB-RAM, over a 1 GByte partition, the kernel is reconfigured to enable the DummyNet options. The DummyNet options used in this emulator are IPFIREWALL, DummyNet, and HZ=1000 to reduce dummynet's processing granularity down to 1ms per task.

Interconnections to end nodes are performed using ethernet configuration. The use of ethernet is similar to the most common systems in the real world. Furthermore, a full duplex fast ethernet is being used to interconnect the end nodes to the WAN Emulator, which will allow the most ideal network performance in every end systems in FANS. Any desired network behavior could be setup within the emulator itself.

5.5.4 Topology implementation in Emulator

Three different configurations will be presented to demonstrate the Emulator's capability to represent the network topology of the desired scenario. These three different configurations are presented starting from the simplest configuration to the more complex ones. Nevertheless, the configurations in this section do not cover all the possibilities that the Emulator could perform. The more sophisticated network behavior could be emulated by utilizing the IPFW command feature such as routing characteristics and some specific interfaces such as loop back interface.

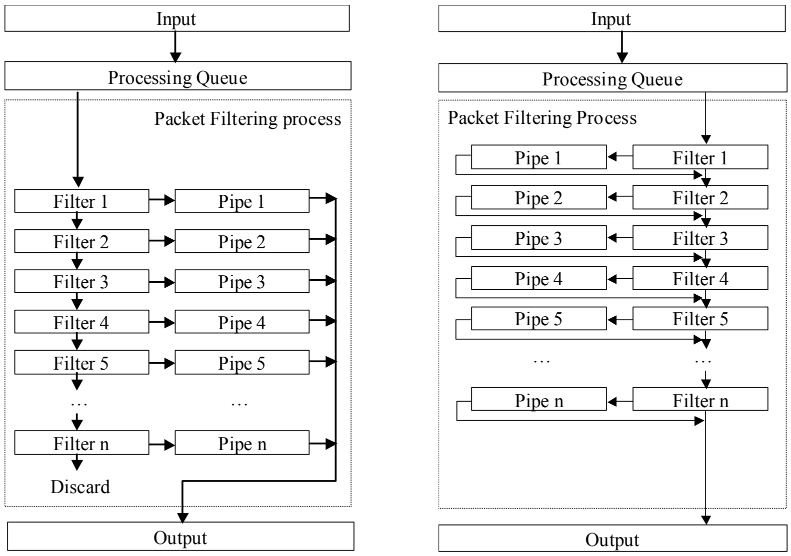


Figure 5.6 Illustration of simple Filter-Pipe mechanism in ipfw+dummynet net.inet.ip.fw.one_pass=1 (links) & net.inet.ip.fw.one_pass=0 (right)

5.5.5 Configuration 1: Network with Separated Links for Ingoing and Outgoing Traffic

A conceptual FANS network incorporates full duplex bi-directional pipe to interconnect endpoints similar to Figure 5.3. According to the topology, there should be a pair of pipe with specific characteristics, i.e., bandwidth, delay and queue size, to emulate the connections between the end nodes. Each pipe of the pair is dedicated to a direction. Figure 5.7 shows the conceptual FANS network structure represented by the pipe constructions.

The *ipfw* configuration of the conceptual FANS is very simple. Every pipe can be configured independently since any traffic among the end nodes only required to be passed through a single pipe in any direction. The packet filtering mechanism thus, can be implemented on each separated IP segment (using CIDR notation) for combinations of source and destination addresses. Furthermore, because of the single pipe-single traffic mechanism, *net.inet.ip.fw.one_pass sysctl* variable is set to 1. APPENDIX G shows the detailed configuration of the conceptual FANS network structure.

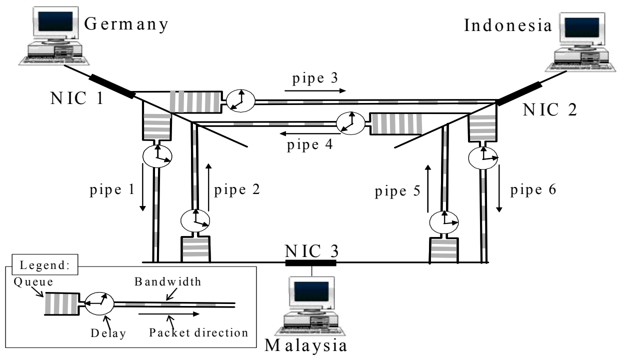


Figure 5.7 Conceptual FANS Network Structure

5.5.6 Configuration 2: Network with Shared Link for Ingoing and Outgoing Traffic

The problem with conceptual FANS network structure is that it is most unlikely for end nodes to maintain separated links for each connection. It is more common to have a single link, which is usually in full duplex connection, to attach the end node or end network with a limited bandwidth and definitely limited queue size to the Internet.

Table 5.1. ICMP measurement over the FANS' end networks

Network 1	Network 2	Average Round trip delay (ms)
Germany: 134.91.0.0/16	Malaysia: 202.185.38.0/24	350 ms
Germany: 134.91.0.0/16	Indonesia: 152.118.0.0/16	831 ms
Indonesia: 152.118.0.0/16	Malaysia: 202.185.38.0/24	720 ms

While the Internet itself is an unpredictable network, simple measurements can be implemented over packet's traffic among the end nodes to observe the specific behavior. Some measurement can be implemented by simply sending some small ICMP packets (64 Bytes) to measure the transmission delay among end nodes. Table 5.1 shows the results of ICMP measurement among the end nodes. Based on the measured round trip delay, a more realistic emulation model can be developed.

Figure 5.8 shows the model's construction called Real World FANS Network. This model represents the round trip delay behavior of WAN over the traffics among the end nodes as well as the bandwidth characteristic of each end network's connection to the Internet similar to the real architecture.

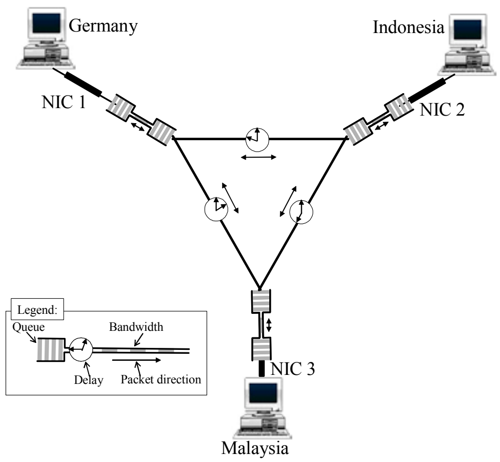


Figure 5.8 Real World FANS Network Structure

The diagram in Figure 5.8 has been simplified to clearly describe the similarity of the emulated network structure to the real architecture. Every symbols used in the diagram is actually constructed by a pair of pipes going in different directions. This will ensure that the traffic will be bidirectional traffic. Compared to network structure in Figure 5.7, the pipes in Figure 5.8 can only be either limited bandwidth without delay (called bandwidth pipes) or unlimited bandwidth with certain amount of delay (called delay pipes). The bandwidth pipes are used to emulate the connection parameters of each link between end nodes to the Internet while that delay pipes are used to emulate the traffic parameter between two end nodes in the Internet.

Since any traffic injected to the emulator should pass exactly three pipes in each direction then the *net.inet.ip.fw.one_pass sysctl* variable should be set to 0. This will allow cascaded pipe mechanism that has been described in Figure 5.6. APPENDIX H shows the detailed configuration of the Real World FANS network structure.

5.5.7 Configuration 3: Network for Measuring Delay Variation

Although the Real World FANS network structure in Section 5.5.6 has represented most of the real world network architecture, it does not emulate the uncertainty of the Internet. The Internet has been known as the best effort network. Although there are efforts to implement some Quality of Service behavior to the Internet, the heterogeneity nature of the Internet would still contributes the most unpredictable performance due to the packet switching and dynamic routing operation in it. Moreover, most of the Internet’s traffics are considered bursty traffic, which will introduce temporal congestion that may cause fluctuation of performance. In terms of traffic’s delay, this means that there will be some delay variation over the transmission time, which will cause some packet sequence reordering by the network.

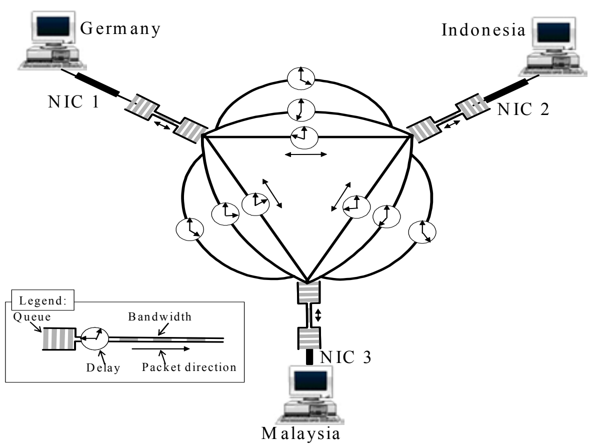


Figure 5.9 The Enhanced Real World FANS Network Structure

In order to emulate the delay variation behavior, a new strategy is implemented. This strategy, which is illustrated in Figure 5.9, utilizes a probabilistic branching mechanism within the delay pipes. Fortunately, the IPFW process in FreeBSD allows such probabilistic multiple path to be implemented by simply adding new parameter in the IPFW command. The parameter has been dealt with the flow of packet filters to implement to the processed traffic. Furthermore, since the packet filters are inspected in sequential manner, some process flow control mechanism can be implemented by using the SKIP TO command (similar to GOTO in most programming language) to jump from a certain point to other filter number/sequence.

Although the number of multiple bi-directional paths shown in Figure 5.9 is 3 (three), this is certainly just a matter of simple delay variation of maximum, mean and minimum value. If desired, other types of delay variation behavior can also be implemented. In this version, the probability of each pipes in the three delay pipes is set evenly to 33% in the way that 33% probability for first pipe occurrence, then 50% of the rest of probability for the second pipe occurrence ((1/2 * (1 – 1/3)) = 1/3), and finally the remaining probability for the third pipe occurrence (1 – 1/3 – 1/3 = 1/3). APPENDIX-I shows the detailed configuration of the Real World FANS network structure.

Chapter Six

FANS Experimentation Results

6.1 Introduction

For experimentation purpose, multimedia applications are streamed over WAN emulator as testbed. Since the testbed emulates the dynamic and unpredictable Internet behaviour such as heterogeneous bandwidth capacities or condition, a variety of transmission delay and routing with multipaths probability, it is hoped that the result is as close as possible to the real world testing. A total of 271 tests have been conducted in the experimentation phase.

Table 6.1 summarizes the testing mode and QoS metrics used to measure the performance of FANS concept and architecture. Information regarding the QoS metrics and testing modes are explained in Chapter Two and Chapter Five respectively.

Table 6.1 Measurement Mode and QoS Metrics used in FANS Benchmarking Test

Measurement Mode	QoS Metrics
In-service test	Response time and frame damage or lost

An important aim of the experiments and performance measurement is to examine the adaptability and reliability of FANS in delivering streaming multimedia application over the heterogeneous bandwidth environment. For reliability testing purpose the performance of direct connection multimedia streaming application (i.e., source – destination) and through intermediate peer with additional post processing (i.e. source - intermediate peer – destination) are compared over the environments whose different bandwidth capacity. The advantages and drawbacks of each approach are further outlined. The adaptability of FANS is investigated by applying, based on the bandwidth capacity of requesting peer and intermediate peer, different codecs configuration. Next, the experimentation is targeted to verify that the enhancement process at a selected intermediate peer can improve the QoS of Multimedia application in particular to reach peers/clients with poor bandwidth. The latest aim of the experiments is to prove that compared to other Internet dynamic factor such delay and multipaths routing, the heterogeneity of bandwidth capacities affects the quality of multimedia applications the most.

6.2 Initial Considerations for Experimentation

In delivering bandwidth-intensive multimedia applications from source node to the requesting client the Internet's traffic condition at the intermediate nodes or hops must be considered. The existence of intermediate hops is apparent in the Internet. In Chapter One it is described that most of the Internet traffic are delivered through USA as one of important hops. The existence and the use of intermediate hop is also the case for FANS. When necessary, a redirected traffic is imposed on the delivery of multimedia application from source to destination. Further, an intermediate peer is selected to perform bandwidth adjustment. If bandwidth capacity of requesting node is sufficient, however, direct delivery is preferred and chosen.

Given its importance, FANS requires that information regarding bandwidth capacity of all peer computers and the size of the shared multimedia files that are available to be announced and distributed to the peer computers. The importance of this information for FANS functionalities is twofold:

(i) To decide if the involvement of intermediate peer is necessary to improve the performance and reliability of the streaming applications

(ii) To select the appropriate intermediate peers among the existing ones

In addition, by imposing this strategy FANS is capable of tracking the members joining and leaving the session, and detecting and determining the location of the multimedia files shared by all the members.

Reflecting the above circumstances and capabilities, the experimentation is carried out based on the following conditions:

1. Bandwidth Information

   Bandwidth capacity of all active FANS participants are announced and distributed through FANS member's tracking mechanism. That is, when a peer computer joins FANS it supplies also information of its bandwidth capacity and the size of the available shared media files. FANS' tracking mechanism updates the information dynamically.
2. Coder and Decoder (codec)

Two types of codecs, which are supported by JMF, namely H263 and JPEG codecs, are used. These codecs are used interchangeably in the experiments to obtain the optimum codecs chain (i.e. source-intermediate-requesting node). The multimedia applications are streamed from server to the requesting node by employing these codecs.

3. Requesting Node

   FANS obtains information about the bandwidth capacity of the requesting node and aboout the size of multimedia file to be streamed, compares them, and decides whether or not the utilization of intermediate peer is necessary to improve its reliability and performance. The size of requested media file, and bandwidth capacity of the requesting node are key factors to determine the necessary type of video quality adjustment.

4. Source node

   FANS uses the information about source node’s bandwidth to transmit the media file using appropriate codec. For a source node with high capacity bandwidth either JPEG or H263 is possible. However this depends also on the bandwidth capacity of the intermediate peer.

5. Intermediate Peer

   For intermediate peer with high bandwidth capacity receiving JPEG-based media file is preferable. This is due to the ability of JPEG to maintain higher quality of motion picture compared to H263. Otherwise, the H263 codec is used.

6.3 Classification of the Quality of Streaming Media

The quality of video is observed based on the number of packet loss and the quality of motion picture presented on the video screen. The classification is derived from the report published by Institute for Telecommunication Science (ITS) for National Telecommunications and Information Administration (NTIA) [WOL90]. It acknowledges two factors that dominantly affect the resolution of streaming multimedia application namely spatial blurring (smeared edges) and temporal blurring (jerky motion).

The information content of a video signal that contains moving and/or changing scenes may simply be too great for Internet client with low bandwidth capacity if no modification is made. The motion artefacts [WOL90], which are most noticeable to the viewer and that show the most potential for being measured are reproduced in Table 6.2.

Table 6.2 Common Video Artefacts [WOL90]

No.	Motion Artefact	Definition
1.	**Resolution Degradation** Examples:	The deterioration of motion video imagery has suffered a loss of such that the received video spatio-temporal resolution.
	Blocking	The received video imagery possesses rectangular or checkerboard patterns not present in the original
	Blurring/smearing	The received video imagery has lost edges and detail present in the original
	Jerkiness	The original smooth and continuous motion is perceived as a series of distinct snapshots
2.	**Edge Busyness** Examples:	The deterioration of motion video objects are displayed with such that the outlines of moving randomly varying activity
	Mosquito noise	The quantizing noise generated by the block processing of moving objects that gives the appearance of false small moving objects (e.g. a mosquito flying around a person's head and shoulders)
3.	**Image Persistence** Example:	The appearance of earlier faded changing object within the current video frames of a moving and/or video frames
	Erasure	An object that was erased continues to appear in the received video imagery

Based on these categorizations and for the purpose of quantifying the experiment result the packet loss is then mapped to the quality of video produced. Next, the streaming video qualities are classified into four groups of qualities as shown in Table 6.3.

Table 6.3 Packet Loss and Quality of Video Presentation

Packet Loss (%)	Quality of Video Resolution
0 – 10	Streaming smoothly, Good Quality
11 – 20	Streaming smoothly, Blurred Image
21 – 30	Jerky motion, Blurred image
> 30	Static Picture. No motion.

Preferring to measure the quality of service in real time fashion, the non-intrusive in-service measurement is used. In this scheme, it is the network itself reports the objective parameter qualities to the requesting peer. This technique can be accomplished with the assistance of JMF real time control protocol (RTCP) inherent mechanism.

6.4 Bandwidth Adaptation through Transcoding and Adjustment of Video Presentation Frame

FANS adapts to the heterogeneity of capacities by comparing the file size and the bandwidth capacity of the requesting node and then taking appropriate modification accordingly. The adaptation is performed on-the-fly by a selected intermediate peer. The light action involves reformatting the media to a smaller format. The heavy response includes scaling down the media presentation frame. The first action produces a less quality of video presentation and the latest causes a certain number of receivers get a smaller media presentation frame depending on their bandwidth capacities.

Table 6.4 Media Formatting and Scaling

Original	Size	New	Codecs	Scale	Size	Saving
AVI	12524	MOV	JPEG	1	11849	5 %
AVI	12524	MOV	JPEG	0.5	2940	86%
AVI	12524	MOV	H263	1	493	96%
AVI	12524	MOV	H263	0.5	492	96%
MOV	11849	MOV	H263	1	746	95%*
MPEG	4583	MOV	JPEG	0.5	7630	No
MPEG	4583	AVI	JPEG	1	8325	No

* converting JPEG to H.263;

Experiments are carried out to verify the effect of reformatting and rescaling multimedia files against their size. As already explained in Chapter Five, the size of video presentation determines the amount of bandwidth a media requires if it is to be electronically transmitted to other places in Internet. Table 6.4 depicts the result of the experiment. While reformatting the transmitted media can reduce the size of media up to 20%, rescaling the media can save up to 96% of the link bandwidth.

Figure 6.1 Full-scale and Half-scale Movies

The media from AVI file is transcoded into other format such as MOV file, and form that new file with JPEG or H263 codecs. For downscaling scheme the 0.5 factor value is used. It means the new file will present movie in half of its original picture size. It is noticed that although transcoding video with H263 codec saves much more bandwidth but the quality of video presentation is worse than transcoding with JPEG codec. Note also that there is no benefit gained from transformatting and scaling MPEG files. The samples depicted in Figure 6.1 show the difference of full-scale and half-scale movies.

6.5 The Effect of Variable Round-trip Delay o the Quality of Media Streaming

Tests are carried out to study the effect of the variety of round-trip delay to the streaming video presentation. The result is shown in Figure 6.2. Both JPEG and H263 codecs are used in this test. Round-trip delays are set from 0 second to 3 seconds, which are the extreme values of the Internet round trip delay. An already-long round trip delay from German to Indonesia, for example, is only 800 ms.

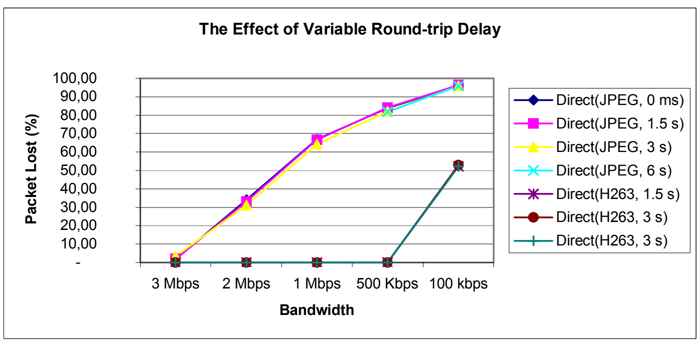


Figure 6.2 Packet Lost vs Round-trip Delay

At round trip delay of 3 seconds there is still no buffer overflow occurred at the router and intermediate node. A buffer overflow is noticed when the round-trip delay is set to 6 seconds and the bandwidth capacity of source node is 3 Mbps. This is a logic result due to limited capacity of the intermediate buffer to cache bandwidth intensive data in addition to create 6 seconds delay in the testbed.

The results show that there is no significant effect of variable round trip delay to the packet lost and the quality of video presentation at the end node. Therefore variety in round trip delay does not affect significantly the quality of streaming multimedia applications over the Internet.

6.6 The Effect of Multiple Paths' Probability on the Quality of Media Streaming

Figure 6.3 depicts the result of experiments to observe the effect of multipaths probability on the quality of streaming media. Media streams from source to destination through three possible paths. Utilizing Configuration 3, each path is weighted accordingly to 0.3/0.35/0.35 and 0.8/0.1/0.1. The tests represent the real condition of Internet where packets might take different routes from source to destination.

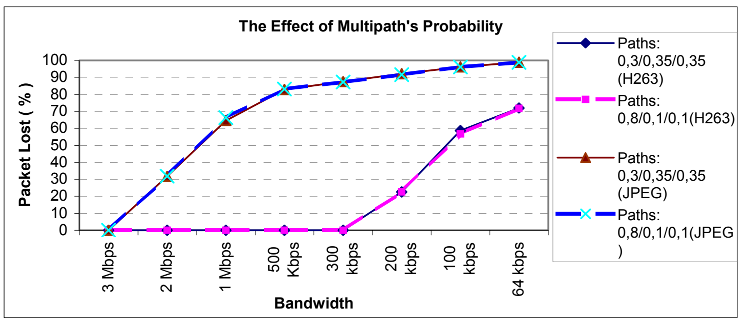


Figure 6.3 The Effect of Multipath Probability on Packet Lost

There is no significant effect of the multipaths probability existed on the Internet to the quality of media streaming and it is true for either JPEG or H263 codecs. The performance of the applications over the different path configurations shown in the above picture is similar.

6.7 The Effect of Heterogeneity of Bandwidth Capacities on the Quality of Media Streaming

One of the hypotheses is that the most significant factor that affects the quality of media streaming over the Internet is the heterogeneity of bandwidth capacities at the client site. Through the following experiments the capability of FANS to adapt to various bandwidth capacities ranging from 5 Mbps or more, to 64 Kbps is examined.

6.7.1 Experiments with H263-H263-H263 Codecs Chain

In this experiment the performance quality of direct connection with H263 codec and transmission of media through intermediate node with H263-H263-H263 codec sequence is compared. The performance is measured by utilizing Configuration 1 and Configuration 2. Figure 6.4 represents the result of the experiments.

The result shows that, by employing H263 codec or a chain of H263 codecs, either direct delivery or indirect delivery, FANS is capable of maintaining zero loss and, hence, is reliable up to 300 kbps. The tolerable performance (i.e., 10% loss), however, is kept stable until 200 kbps. Slightly different performance in favour of FANS reroute approach, though not really beneficial due to the already high loss, appears at 100 kbps.

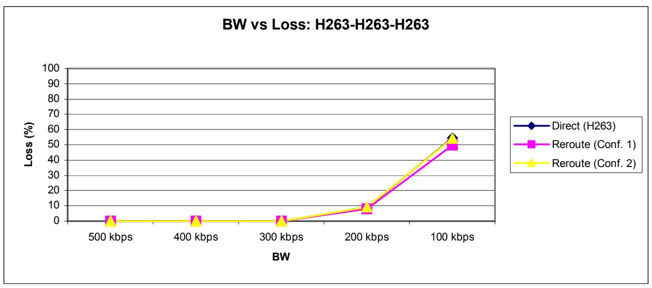


Figure 6.4 BW vs Loss: H263-H263-H263

To conclude, the performance of FANS reroute mechanism is similar with the direct connection. The quality of video presentation at the requesting peer is fair. This is due to the use of H263 codec.

6.7.2 Experiments with H263-H263-JPEG Codecs Chain

Figure 6.5 shows the performance comparison of direct connection with JPEG and indirect connection with H263-H263-JPEG chain. H263 codec is used from media source to intermediate node to maintain 100% reception. At the intermediate node media file is saved with H263 codec to keep its size small. Bottleneck problem occurs at the final phase from intermediate node to requesting peer). The effect output of using JPEG codec is apparent with the big amount of packet loss when the bandwidth capacity is low.

Figure 6.5 BW vs Loss: H263-H263-JPEG

Direct connection maintains zero loss at 3 Mbps or higher. FANS improves the performance by keeping the tolerable performance (i.e., 10% loss) up to 2 Mbps. The presentation quality for direct connection is better than indirect connection with FANS. This is due to the involvement of H263 within the codecs chain, which causes the degradation of resolution.

6.7.3 Experiments with H263-JPEG-H263 Codecs Chain

In these experiments, the intermediate node receives the media with JPEG codec and transfers it to origin with H263 codec. The performance of FANS (i.e., indirect route) is superior to the direct route. This is depicted in Figure 6.6. FANS maintains the level of packet loss to zero even at the band-width of 100 Kbps, while the direct connection keeps the tolerable performance only up to 200 Kbps.

Figure 6.6 BW vs Loss: H263-JPEG-H263

However, the quality of picture produced by both approaches is fair. This is due to the use of H263 codec at source and from intermediate peer to requesting peer.

6.7.4 Experiments with H263-JPEG-JPEG Codecs Chain

Knowing that JPEG codec maintains the quality of video presentation closest to the original the use of JPEG codec during transmission is increased. A better quality of video presentation is achieved but with much bigger loss. As a result, video motion is broken and static. The results of the experiment are depicted in Figure 6.7.

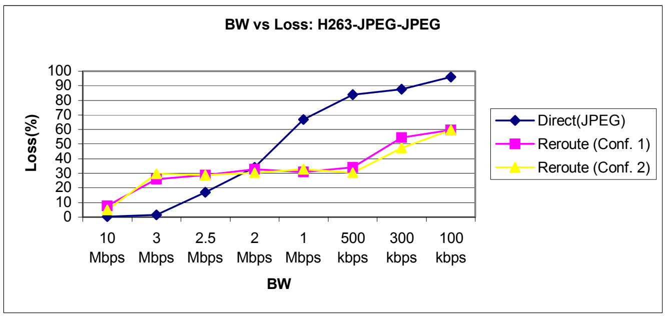


Figure 6.7 BW vs Loss: H263-JPEG-JPEG

Note that the FANS reroute mechanism is compared with direct transmission with JPEG codec, and not with H263 codec. Employing JPEG codec, both direct connection and FANS approach require high capacity bandwidth to perform sufficiently.

6.7.5 Experiments with JPEG-JPEG-JPEG Codecs Chain

In this experiment, the demand of high capacity of bandwidth by utilizing JPEG codec in all phase of the media transmission is extremely increased. The expected outcome is that the quality of video presentation to be as good as the original one throughout the nodes.

However this is not a good option since it works well only when all the parties involved have minimum bandwidth of 5 Mbps or even 10 Mbps. Figure 6.8 shows the outcomes. This case, in addition, is not a likely case to properly represent heterogeneity nature of bandwidth capacities in Internet.

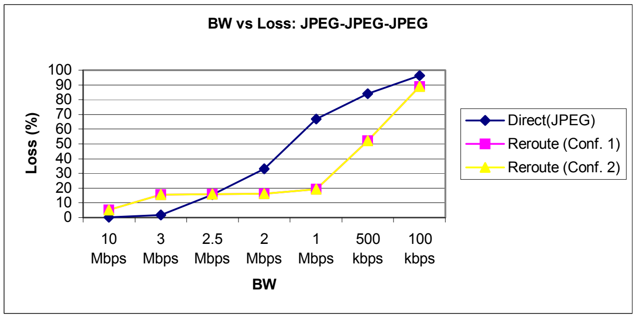


Figure 6.8 BW vs Loss: JPEG-JPEG-JPEG

The loss of packet is very high starting at 1 Mbps. The movie is static. That is, no motion can be seen. Overall performance shows that, although FANS is not really successful in reducing packet loss for clients with poor bandwidth, it performs better at lower bandwidth region between 2.5 Mbps and 500 Kbps.

6.7.6 Experiments with JPEG-JPEG-H263 Codecs Chain

With this combination, the quality of video presentation is maintained as good as possible until the intermediate node. The assumption and precondition is that the intermediate has least similar the bandwidth capacity to media source. At the final stage H263 codec is used to reduce bandwidth demand and to adapt to low bandwidth capacity of the requesting peer. The results of experiments are represented in Figure 6.9.

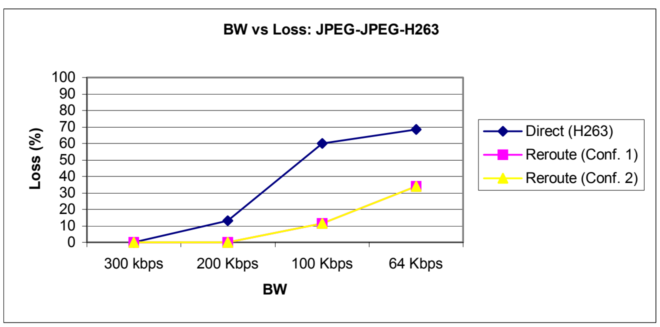


Figure 6.9 BW vs Loss: JPEG-JPEG-H263

FANS successfully maintains the quality of picture to be fair, similar to direct transmission with H263 codec. Moreover, FANS improves the reliability of media transmission by reducing the number of lost packet. In short, FANS performs more reliably than direct connection by keeping tolerable performance up to 100 Kbp

6.8 FANS Overall Performance

Figure 6.10 represents the overall result of the experiments. There are three combinations of FANS reroute mechanism and chain of codecs, i.e. H263-H263-H263, H263-JPEG-H263 and JPEG-JPEG-H263, that overpower the direct connection either with JPEG or H263 codec in terms of the reliability of the streaming media. Out of these three combinations JPEG-JPEG-H263 order offers the best quality of video presentation.

With JPEG-JPEG-H263 order the system maintains the quality of video presentation as good as possible until the intermediate peer and opts for the efficient mode, i.e. H263 codec, for the final stage to adapt to the low capacity of the client's bandwidth. However, streaming media with JPEG codec at intermediate peer as is the case with JPEG-JPEG-H263 demands that the peer has high capacity of bandwidth.

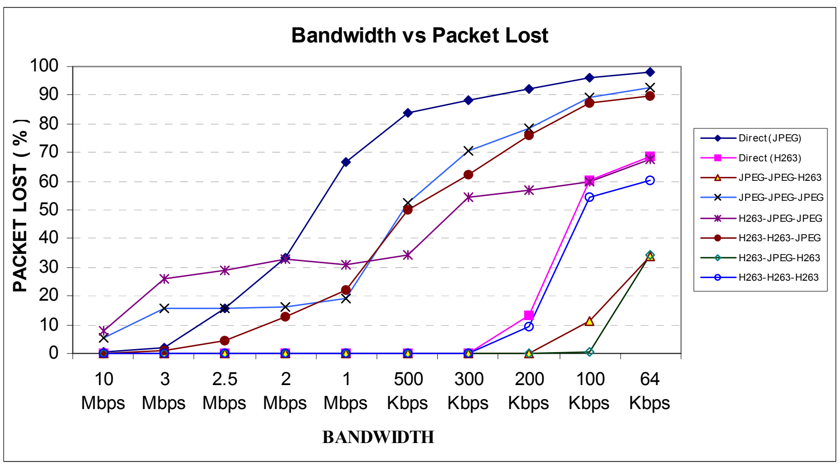


Figure 6.10 Bandwidth Heterogeneity vs Packet Lost

H263 codec requires only 300 Kbps of bandwidth to stream smoothly. Therefore although produces less quality of video presentation H263 codec should always be employed in the chain of codec combination if either source node or intermediate peer or requesting node has only at most 300

Kbps bandwidth. For that reason it is noticed also that the JPEG-H263-H263 is the most reliable codecs sequence to maintain packet lost in an acceptable manner if no other nodes but the source node having high bandwidth capacity. Due to the less than fair quality of video presentation it produces, the combination of H263-H263-H263 is only to be considered if the requesting node has less bandwidth than 100 Kbps.

6.9 Weighing FANS against RealPlayer

An advanced test is conducted by weighing FANS against a commercial product namely RealOne@ Player. Due to some incompatibilities between both systems the following conditions for RealOne are noticed:

1. RealOne, which is a newest version of RealPlayer, does not support H263 codecs and MOV format.
2. The level of video picture quality is as original, i.e., maintains its best. No presentation frame resizing or data codecs are applied.
3. HelixServer is used to support RTSP streaming from server to client in the testbed

On the contrary, FANS which uses JMF as its video processor, does not support RealMedia (i.e., *.rm) format which is the most widely supported and the fastest format running over RealPlayer tools. Therefore the format used in the performance test is the one which is supported by both FANS and RealOne, and, that is, AVI format.

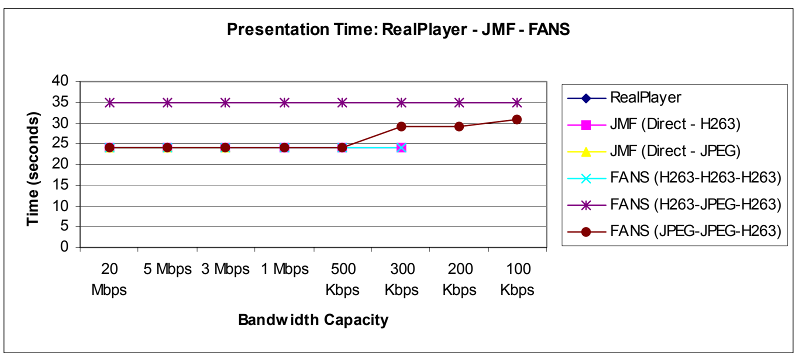


Figure 6.11 Result of the Performance Test on Presentation Time

It is noted that the RealOne is capable of maintaining the smooth and perfect video transmission and presentation at the minimum bandwidth of 5

Mbps at the client side, which is a very demanding one. Figure 6.11 depicts the result of this test in terms of presentation time. For JMF (Direct-JPEG) the video runs smoothly until 3 Mbps and JMF (Direct-H253) performs reliably until 300 Kbps. FANS mechanism with the combination of H263-JPEG-H263 and JPEG-JPEG-H263 are the most reliable one and can reach as low as 128 Kbps, even though with some additional delay due to rebuffering and A/V synchronizing processes at the client side.

Figure 6.12 represents the result for the transmission time, it is noted that for RealPlayer, starting at the minimum of 3 Mbps bandwidth capacity, the video presentation and transmission stop at the 13rd second and that the voice continuously streaming again after until finish the 72nd second. At the lower bandwidth, the video stop streaming constantly at the 13rd second but the delay at which the voice starts streaming again is getting worst. That is, after 126th second at 1 Mbps, after 239th second at 500 Kbps, after 416th second at 300 Kbps, after 580th second at 200 Kbps and after 1114th second at 100 Kbps. Other technique such as JMF and FANS performs well with various delays, in this experiment. Therefore based on this test, for multimedia file of AVI format, FANS performs much better than RealOne in a heterogeneous bandwidth environment.

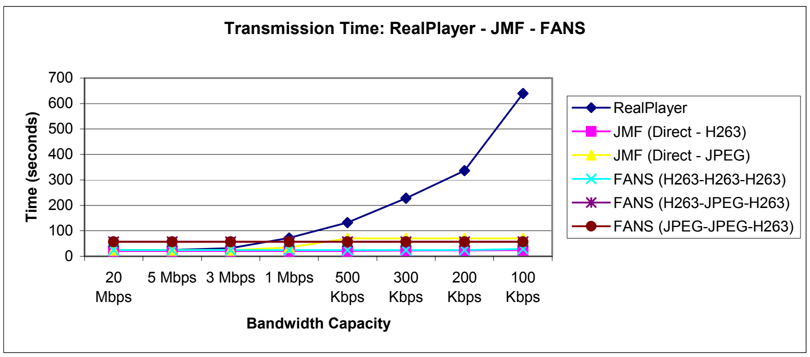


Figure 6.12 Result of Performance Test on Transmission Time

6.10 FANS Improvement Phase: Reducing Setup Delay

The previous FANS experiments show that the transmission delay for redirection path (i.e., averaged 83 secs) is much longer than direct connection (i.e., averaged 47 secs) by 76 percent. Most of the delay is due to the intermediate peer's post-processing overhead. The intermediate peer's post-processing was expensive in terms of time because FANS intermediate system has to decode the media it receives based on the enhancement format and save it to a file. FANS intermediate peer then read again the file to transmit it to the requesting node. The function of FANS intermediate peer is successfully modified to support transcoding and rescaling purpose only. This new strategy eliminates the need for storing the media to temporary file. With this mechanism FANS decreases the setup delay to 60 seconds. Although the total setup delay is still 28 percent longer than direct connection, this is worth of 38 percent improvement compared to the previous approach.

The comparison is given in Table 6.5. Nonetheless, this fact supports the correctness of the design consideration phase to selectively perform intermediate enhancement only if really necessary. That is, when streaming the media to clients with poor bandwidth.

Table 6.5 FANS Delay Comparison

Type of Connections	Delay (s)	Difference (%)
Direct Connections	47	--
Redirect Connections (with intermediate file)	83	76
Redirect Connections (no intermediate file)	60	28

6.11 Notes on Audio/Video Synchronization Problem

FANS architecture relies on JMF technology. It is noticed that FANS is still experiencing audio/video synchronization problem. At the destination peer the video leads the audio in miliseconds. The first prediction is that it is JMF internal problem since the problem exists not only in FANS redirecting scheme, but also for direct media delivery. However, this might also be related to Java virtual machine concurrent scheduling issues.

Although communication and support from JMF engineering team and JMF community has been obtained to solve this problem, effort to get rid of it has been unsuccessful. The comments of JMF engineering team and community regarding this issue are attached in APPENDIX-K. Therefore the problem is left for nearest future enhancement. The proposed solution includes considering other JMF-like APIs, or developing FANS own video presentation and transcoding tools.

Chapter Seven

Conclusion and Future Issues

7.1 Conlucsion

1. The improved active network architecture promotes fairness in terms of bandwidth diversity among peers. Peers with high bandwidth capacities obtain higher quality of multimedia application than peers whose poor bandwidth capacities.
2. The improved active network architecture is capable of streaming multimedia applications over environment with heterogeneous bandwidth capacities like the Internet. In the experiment, it is proven that for 12 MB multimedia data, the system is capable of supporting the heterogeneity of bandwidth capacities ranging from 3 Mbps to 128 Kbps. This is a significant improvement compared to similar work by [YAM02] which uses router level active networks and is capable of supporting bandwidth capacities as lowest as 2 Mbps for 8 MB multimedia data.
3. The involvement of intermediate enhancement processing must be carried out selectively due to its expensiveness in latency. The improved active network architecture supports this strategy. The experiment shows as well that traffic adaptation at intermediate peer improves the reliability of transmission, particularly in accommodating peers with poor bandwidth.
4. From the experiment, it is shown that BW is the dominant factor affecting the end-to-end quality of streaming applications. It is also noticed that 10% packet loss is acceptable for human vision watching streaming multimedia applications.
5. The improved active network architecture is capable of supporting the mobility and dynamic existence of peers. The tracking mechanism and algorithm facilitate this feature.
6. Take advantage of application level active network paradigm the improved active network architecture is applicable in current Internet condition. An important advantage of this approach is that it requires no

modification in current Internet infrastructure and does not add complexity to router level management.

7.2 Future Issues and Related Research Fields

FANS tracking mechanism depends on the functionality of service registration point (SRP). All peers contact SRP on regular basis to get new update information if any. Addition, deletion and editing occurs at local peer are reported to SRP. SRP is then responsible to maintain the list of member and media correctly. Once a peer contacts SRP, SRP transfers latest condition on lists to the peer. This is a kind of centralized management. The problem will occur when the number of member and media increase significantly and the SRP get overloaded to handle the exchange of information. Therefore FANS will need decentralized tracking strategy and is subject for future work. The decentralization approach will determine further the level of scalability that will be supported by FANS in the future.

Another problem related to the tracking mechanism is “lost update”. It is possible that reports from peer could not reach the SRP, or two updates appear at exactly the same time to the shared lists and lost update occurs. Consequently the lists maintained by SRP are not the correct ones. Further, this will affect “the knowledge” possesses by peers regarding the existence of shared media and influence the intermediate peer’s selection objective. Synchronization and update strategies should be developed and implemented to address this issue.

Future work is also necessary to address the issue of load balancing problem. The current selection mechanism selects intermediate peer randomly and prefers to select intermediate peer with high bandwidth capacity. With this strategy, it is possible an intermediate peer with high bandwidth capacity is selected many times whereas the other peers, with middle bandwidth capacity but are sufficient to perform intermediate peer’s function in FANS, are idle. Load balancing of Internet services is still an open research field although some strategies exist already.

The concept of peer computers in the architecture is inherited from similar concept in Grid computing notion, where peer computers are those individuals and organizations, which agree to form a virtual organization and share their resources for common purpose. The assumption is that the members of this virtual organization are known and trustworthy. Therefore current FANS does not address security issue. However, when the architecture concept and system grow into maturity, it will be possible to include peer computers from wider parties and public area. In this condition, security is serious issue and should be addressed thoroughly.

Other application areas for the system concept and architecture that should be considered include ad-hoc mobile network and pervasive computing. FANS system architecture covers all the functionalities required for these applications: bandwidth adaptability, mobility support, device's tracking mechanism, and communication and media transfer capabilities. Further research gates are wide opened in these relatively new network computing fields.

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APPENDIX – A

Internet Traffic Trace: Germany – USA

Target: www.mit.edu Date: 11.03.2003 (Tuesday), 20:35:39 Nodes: 20

Node Data					
Node	**Net**	**Reg**	**IP Address**	**Location**	**Node Name**
1	-	-	192.168.192.3	Duisburg	Clienta
2	1	-	192.168.192.1	-	
3	1	-	192.168.64.2	-	
4	1	-	192.168.80.2	-	
5	1	-	192.168.82.1	-	
6	2	1	134.91.90.1	-	Cisbae1.uni-duisburg.de
7	2	1	134.91.3.3	-	routle-hrz.uni-duisburg.de
8	2	1	134.91.254.2	-	msfc65le.uni-duisburg.de
9	3	2	188.1.44.1	Krefeld	ar-essen1.g-win.dfn.de
10	3	2	188.1.86.1	Krefeld	cr-essen1.g-win.dfn.de
11	3	2	188.1.18.89	Wiesbaden	cr-frankfurt1.g-win.dfn.de
12	3	2	188.1.80.38	Wiesbaden	ir-frankfurt2.g-win.dfn.de
13	4	3	62.40.103.33	-	dfn.de1.de.geant.net
14	4	3	62.40.96.50	-	de.fr1.fr.geant.net
15	4	3	62.40.96.90	-	fr.uk1.uk.geant.net
16	4	3	62.40.103.26	-	abilene-gw.uk1.uk.geant.net
17	5	4	192.5.89.9	-	atm10-420-oc12-gigapopne.nox.org
18	5	-	192.5.89.90	-	
19	6	5	18.168.0.21	Worcester	nw12-rtr-2-backbone.mit.edu
20	6	5	18.181.0.31	Worcester	dandelion-patch.mit.edu

Network Data

Network id#:1
```
OrgName:    Internet Assigned Numbers Authority
OrgID:      IANA
Address:    4676 Admiralty Way, Suite 330
City:       Marina del Rey
StateProv:  CA
PostalCode: 90292-6695
Country:    US
```

Network id#:2
```
OrgName:    University of Duisburg
OrgID:      UNIVER-283
Address:    Lotharstrasse 65
City:       Duisburg, D-47048
StateProv:
PostalCode:
Country:    DE
```

Network id#:3
```
OrgName:    RIPE Network Coordination Centre
OrgID:      RIPE
Address:    Singel 258
Address:    1016 AB
City:       Amsterdam
StateProv:
PostalCode:
Country:    NL
```

Network id#:4
```
Francis House, 112 Hills Road
   Cambridge CB2 1PQ, UK
```

Network id#:5
```
OrgName:    Harvard University
OrgID:      HARVAR
Address:    Network Services Division - Office for Information Technology
Address:    10 Ware Street
City:       Cambridge
StateProv:  MA
PostalCode: 02138
Country:    US
```

Network id#:6
```
OrgName:    Massachusetts Institute of Technology
OrgID:      MIT-2
Address:    Laboratory for Computer Science
Address:     545 Main Street
City:       Cambridge
StateProv:  MA
PostalCode: 02139
Country:    US
```

APPENDIX – B

Internet Traffic Trace : Germany - Australia

Target: www.monash.edu.au Date: 11.03.2003 (Tuesday), 20:41:33 Nodes: 29

Node Data					
Node	**Net**	**Reg**	**IP Address**	**Location**	**Node Name**
1	-	-	192.168.192.3	Duisburg	clienta
2	1	-	192.168.192.1	-	
3	1	-	192.168.64.2	-	
4	1	-	192.168.80.2	-	
5	1	-	192.168.82.1	-	
6	2	1	134.91.90.1	-	cisbae1.uni-duisburg.de
7	2	1	134.91.3.3	-	routle-hrz.uni-duisburg.de
8	2	1	134.91.254.2	-	msfc65le.uni-duisburg.de
9	3	2	188.1.44.1	Krefeld	ar-essen1.g-win.dfn.de
10	3	2	188.1.86.1	Krefeld	cr-essen1.g-win.dfn.de
11	3	2	188.1.18.89	Wiesbaden	cr-frankfurt1.g-win.dfn.de
12	3	2	188.1.80.38	Wiesbaden	ir-frankfurt2.g-win.dfn.de
13	4	3	62.40.103.33	-	dfn.de1.de.geant.net
14	4	3	62.40.96.50	-	de.fr1.fr.geant.net
15	4	3	62.40.96.90	-	fr.uk1.uk.geant.net
16	4	3	62.40.103.26	-	abilene-gw.uk1.uk.geant.net
17	5	4	198.32.8.82	-	chinng-nycmng.abilene.ucaid.edu
18	5	4	198.32.8.77	-	iplsng-chinng.abilene.ucaid.edu
19	5	4	198.32.8.81	-	kscyng-iplsng.abilene.ucaid.edu
20	5	-	198.32.8.13	-	

Node Data

21	5	4	198.32.11.110	-	dnvrng-dnvr.abilene.ucaid.edu
22	5	4	198.32.8.49	-	sttlng-dnvrng.abilene.ucaid.edu
23	5	5	198.32.170.45	-	aarnet-pwave.pnw-gigapop.net
24	6	6	192.231.212.33	-	pos1-0.sccn1.broadway.aarnet.net.au
25	6	7	192.231.212.18	-	nswrno2-gbe10-0-0-916.nswrno.net.au
26	7	8	192.12.76.2	-	nsw-vic.atm.net.aarnet.edu.au
27	8	-	203.21.130.40	-	monash-gw1.vrn.edu.au
28	-	-	130.194.28.12	Melbourne	clay1-gw-28.net.monash.edu.au
29	9	-	130.194.11.4	Melbourne	www.monash.edu.au

Network Data

Network id#:1
```
OrgName:    Internet Assigned Numbers Authority
OrgID:      IANA
Address:    4676 Admiralty Way, Suite 330
City:       Marina del Rey
StateProv:  CA
PostalCode: 90292-6695
Country:    US
```

Network id#:2
```
OrgName:    University of Duisburg
OrgID:      UNIVER-283
Address:    Lotharstrasse 65
Address:    Duisburg, D-47048
City:
StateProv:
PostalCode:
Country:    DE
```

Network id#:3
```
OrgName:    RIPE Network Coordination Centre
OrgID:      RIPE
Address:    Singel 258
Address:    1016 AB
City:       Amsterdam
StateProv:
PostalCode:
Country:    NL
```

Network id#:4
```
Francis House, 112 Hills Road
   Cambridge CB2 1PQ, UK
```

Network id#:5
```
OrgName:    Exchange Point Blocks
OrgID:      EPB
Address:    PO 12317
City:       Marina del Rey
StateProv:  CA
PostalCode:
Country:    US
```

Network id#:6
```
OrgName:    CSIRO IT Services
OrgID:      CIS-56
Address:    PO Box 225 Dickson
```

Network Data

```
Address:    Canberra, ACT 2602
City:
StateProv:
PostalCode:
Country:    AU
```

Network id#:7

```
OrgName:    University of Queensland
OrgID:      UNIVER-237
Address:    Brisbane, QLD 4072
City:
StateProv:
PostalCode:
Country:    AU
```

Network id#:8

```
The University of Melbourne
   800 Swanston Street
   Parkville
   VIC 3052
```

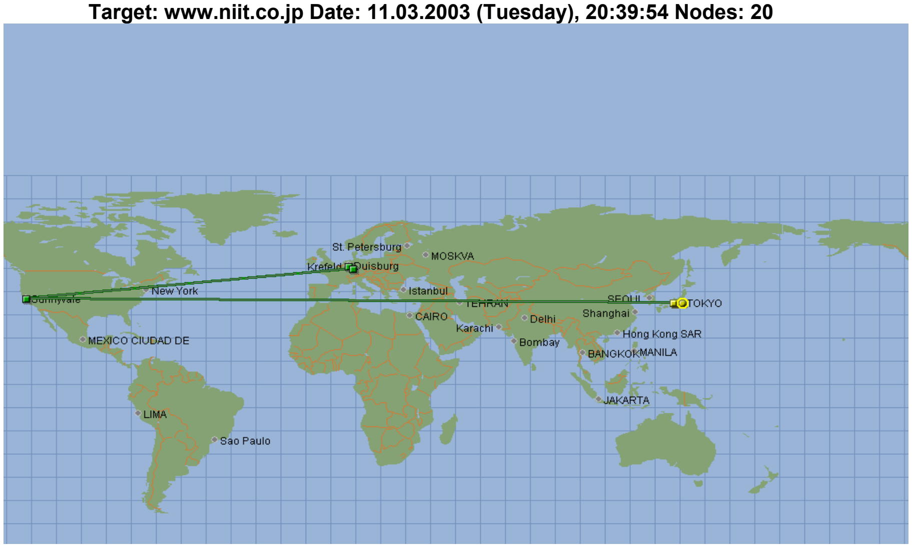
Network id#:9

```
OrgName:    Monash University
OrgID:      MONASH
Address:    IT Facilities  Wellington Road
Address:    Clayton, Vic, 3168
City:
StateProv:
PostalCode:
Country:    AU
```

APPENDIX – C

Internet Traffic Trace: Germany - Japan

Target: www.niit.co.jp Date: 11.03.2003 (Tuesday), 20:39:54 Nodes: 20



Node Data						
Node	**Net**	**Reg**	**IP Address**	**Location**		**Node Name**
1	-	-	192.168.192.3	Duisburg		clienta
2	1	-	192.168.192.1	-		
3	1	-	192.168.64.2	-		
4	1	-	192.168.80.2	-		
5	1	-	192.168.82.1	-		
6	2	1	134.91.90.1	-		cisbae1.uni-duisburg.de
7	2	1	134.91.3.3	-		routle-hrz.uni-duisburg.de
8	2	1	134.91.254.2	-		msfc65le.uni-duisburg.de
9	3	2	188.1.44.1	Krefeld		ar-essen1.g-win.dfn.de
10	3	2	188.1.86.1	Krefeld		cr-essen1.g-win.dfn.de
11	3	2	188.1.18.89	Wiesbaden		cr-frankfurt1.g-win.dfn.de
12	4	3	208.48.23.141	-		so-6-0-0.ar2.fra2.gblx.net
13	5	3	62.16.32.77	-		pos5-0-2488m.cr2.fra2.gblx.net
14	6	-	64.211.147.86	-		
15	7	3	207.136.163.126	-		so6-0-0-2488m.br2.pao2.gblx.net
16	8	-	208.50.13.126	-		
17	9	4	216.98.96.179	Sunnyvale		paloalto-bb3.iij.net
18	10	4	216.98.96.210	35.725N	136.383E	otemachi-bb8.iij.net
19	11	4	210.130.130.22	35.725N	136.383E	otemachi-gate07.iij.net
20	11	5	210.138.141.66	TOKYO		www.niit.co.jp

Network Data

Network id#:1
```
OrgName:    Internet Assigned Numbers Authority
OrgID:      IANA
Address:    4676 Admiralty Way, Suite 330
City:       Marina del Rey
StateProv:  CA
PostalCode: 90292-6695
Country:    US
```

Network id#:2
```
OrgName:    University of Duisburg
OrgID:      UNIVER-283
Address:    Lotharstrasse 65
Address:    Duisburg, D-47048
City:
StateProv:
PostalCode:
Country:    DE
```

Network id#:3
```
OrgName:    RIPE Network Coordination Centre
OrgID:      RIPE
Address:    Singel 258
Address:    1016 AB
City:       Amsterdam
StateProv:
PostalCode:
Country:    NL
```

Network id#:4
```
Global Crossing GBLX-6A (NET-208-48-0-0-1)
                                  208.48.0.0 - 208.48.63.255
GC Internal FGC-REQ000000004336 (NET-208-48-23-0-1)
                                  208.48.23.0 - 208.48.23.255
```

Network id#:5
```
Global Crossing Telecommunications, Inc.
   Olympia 6
   1213 NP Hilversum
   The Netherlands
```

Network id#:6
```
Global Crossing GBLX-11C (NET-64-211-0-0-1)
                                  64.211.0.0 - 64.211.223.255
GC Internal Department FGC-REQ000000008652 (NET-64-211-147-0-1)
                                  64.211.147.0 - 64.211.147.255
```

Network id#:7
```
OrgName:    Global Crossing
OrgID:      GBLX
Address:    14605 South 50th Street
City:       Phoenix
StateProv:  AZ
PostalCode: 85044-6471
Country:    US
```

Network id#:8
```
Global Crossing GBLX-6C (NET-208-48-224-0-1)
                                  208.48.224.0 - 208.50.127.255
```

Network Data

```
GC Internal FGC-REQ000000005067 (NET-208-50-13-0-1)
                                  208.50.13.0 - 208.50.13.255
```

Network id#:9

```
IIJ America, Inc. IIJ-AMERICA-NET (NET-216-98-96-0-1)
                                  216.98.96.0 - 216.98.127.255
IIJ America, Inc. IIJA-96 (NET-216-98-96-0-2)
                                  216.98.96.0 - 216.98.96.255
```

Network id#:10

```
IIJ America, Inc. IIJ-AMERICA-NET (NET-216-98-96-0-1)
                                  216.98.96.0 - 216.98.127.255
IIJ America, Inc. IIJA-96 (NET-216-98-96-0-2)
                                  216.98.96.0 - 216.98.96.255
```

Network id#:11

```
Kokusai-Kougyou-Kanda Bldg 6F, 2-3-4 Uchi-Kanda
   Chiyoda-ku, Tokyo 101-0047, Japan
```

APPENDIX – D

Internet Traffic Trace: Germany - Singapore

Target: www.nus.edu.sg Date: 11.03.2003 (Tuesday), 20:37:36 Nodes: 25

Node Data					
Node	**Net**	**Reg**	**IP Address**	**Location**	**Node Name**
1	-	-	192.168.192.3	Duisburg	clienta
2	1	-	192.168.192.1	-	
3	1	-	192.168.64.2	-	
4	1	-	192.168.80.2	-	
5	1	-	192.168.82.1	-	
6	2	1	134.91.90.1	-	cisbae1.uni-duisburg.de
7	2	1	134.91.3.3	-	routle-hrz.uni-duisburg.de
8	2	1	134.91.254.2	-	msfc65le.uni-duisburg.de
9	3	2	188.1.44.1	Krefeld	ar-essen1.g-win.dfn.de
10	3	2	188.1.86.1	Krefeld	cr-essen1.g-win.dfn.de
11	3	2	188.1.18.89	Wiesbaden	cr-frankfurt1.g-win.dfn.de
12	3	2	188.1.80.38	Wiesbaden	ir-frankfurt2.g-win.dfn.de
13	4	3	62.40.103.33	-	dfn.de1.de.geant.net
14	4	3	62.40.96.50	-	de.fr1.fr.geant.net
15	4	3	62.40.96.90	-	fr.uk1.uk.geant.net
16	4	3	62.40.103.26	-	abilene-gw.uk1.uk.geant.net
17	5	4	198.32.8.82	-	chinng-nycmng.abilene.ucaid.edu
18	5	4	198.32.8.77	-	iplsng-chinng.abilene.ucaid.edu
19	5	4	198.32.8.81	-	kscyng-iplsng.abilene.ucaid.edu

Node Data					
20	5	4	198.32.8.102	-	snvang-kscyng.abilene.ucaid.edu
21	6	5	202.8.94.33	-	abilene-pos-gw3.singaren.net.sg
22	6	5	202.8.94.61	-	gw3-pos-gw1.singaren.net.sg
23	6	5	202.8.94.2	-	gw1-nus.singaren.net.sg
24	7	6	137.132.12.124	SINGAPORE	nusinfo.nus.edu.sg
25	7	6	137.132.12.124	SINGAPORE	nusinfo.nus.edu.sg

Network Data

Network id#:1
```
OrgName:    Internet Assigned Numbers Authority
OrgID:      IANA
Address:    4676 Admiralty Way, Suite 330
City:       Marina del Rey
StateProv:  CA
PostalCode: 90292-6695
Country:    US
```

Network id#:2
```
OrgName:    University of Duisburg
OrgID:      UNIVER-283
Address:    Lotharstrasse 65
Address:    Duisburg, D-47048
City:
StateProv:
PostalCode:
Country:    DE
```

Network id#:3
```
OrgName:    RIPE Network Coordination Centre
OrgID:      RIPE
Address:    Singel 258
Address:    1016 AB
City:       Amsterdam
StateProv:
PostalCode:
Country:    NL
```

Network id#:4
```
Francis House, 112 Hills Road
   Cambridge CB2 1PQ, UK
```

Network id#:5
```
OrgName:    Exchange Point Blocks
OrgID:      EPB
Address:    PO 12317
City:       Marina del Rey
StateProv:  CA
PostalCode:
Country:    US
```

Network id#:6
```
Computer Centre
   2 Engineering Drive 4
   National University of Singapore
   Singapore 117584
```

Network id#:7
```
OrgName:    National University of Singapore
```

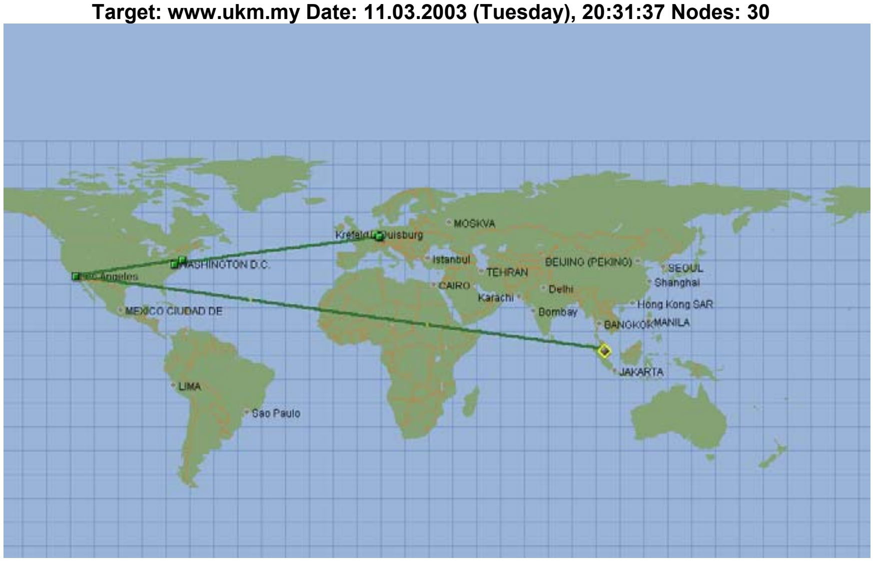
Network Data

```
OrgID:       NUS-1
Address:     2 Engineering Drive 4, National
Address:     University of Singapore117584
City:
StateProv:
PostalCode:
Country:     SG
```

APPENDIX – E

Internet Traffic Trace: Germany - Malaysia

Target: www.ukm.my Date: 11.03.2003 (Tuesday), 20:31:37 Nodes: 30



Node Data					
Node	**Net**	**Reg**	**IP Address**	**Location**	**Node Name**
1	-	-	192.168.192.3	Duisburg	clienta
2	1	-	192.168.192.1	-	
3	1	-	192.168.64.2	-	
4	1	-	192.168.80.2	-	
5	1	-	192.168.82.1	-	
6	2	1	134.91.90.1	-	cisbae1.uni-duisburg.de
7	2	1	134.91.3.3	-	routle-hrz.uni-duisburg.de
8	2	1	134.91.254.2	-	msfc65le.uni-duisburg.de
9	3	2	188.1.44.1	Krefeld	ar-essen1.g-win.dfn.de
10	3	2	188.1.86.1	Krefeld	cr-essen1.g-win.dfn.de
11	3	2	188.1.18.89	Wiesbaden	cr-frankfurt1.g-win.dfn.de
12	4	3	208.48.23.141	-	so-6-0-0.ar2.fra2.gblx.net
13	5	3	62.16.32.77	-	pos5-0-2488m.cr2.fra2.gblx.net
14	6	3	208.178.174.78	-	pos0-0-622m.cr1.wdc2.gblx.net
15	7	3	64.214.65.141	-	so1-2-0-2488m.ar1.dca3.gblx.net
16	8	-	208.51.74.14	-	
17	9	4	207.45.223.1	WASHINGTON D.C.	if-10-0.core1.washington.teleglobe.net
18	9	4	207.45.223.122	WASHINGTON D.C.	if-4-0.core2.washington.teleglobe.net

Node Data

19	10	4	64.86.83.213	-			if-2-0.core2.newark.teleglobe.net
20	10	4	64.86.83.165	New York			if-1-0.core3.newyork.teleglobe.net
21	10	4	64.86.83.174	Los Angeles			if-8-0.core2.losangeles.teleglobe.net
22	9	4	207.45.223.62	Los Angeles			if-5-0.core1.losangeles.teleglobe.net
23	11	4	66.110.10.10	Los Angeles			ix-7-3.core1.losangeles.teleglobe.net
24	12	-	61.6.17.33	-			pos3-0.mea90.jaring.my
25	12	-	61.6.17.45	-			
26	13	-	161.142.25.85	3.125N		101.717E	pos2-0.bkj90.jaring.my
27	13	-	161.142.100.2	-			
28	13	-	161.142.78.22	-			
29	13	-	161.142.34.78	-			
30	14	-	202.185.42.65	3.130N		101.720E	www.ukm.my

Network Data

Network id#:1
```
OrgName:    Internet Assigned Numbers Authority
OrgID:      IANA
Address:    4676 Admiralty Way, Suite 330
City:       Marina del Rey
StateProv:  CA
PostalCode: 90292-6695
Country:    US
```

Network id#:2
```
OrgName:    University of Duisburg
OrgID:      UNIVER-283
Address:    Lotharstrasse 65
Address:    Duisburg, D-47048
City:
StateProv:
PostalCode:
Country:    DE
```

Network id#:3
```
OrgName:    RIPE Network Coordination Centre
OrgID:      RIPE
Address:    Singel 258
Address:    1016 AB
City:       Amsterdam
StateProv:
PostalCode:
Country:    NL
```

Network id#:4
```
Global Crossing GBLX-6A (NET-208-48-0-0-1)
                                  208.48.0.0 - 208.48.63.255
GC Internal FGC-REQ000000004336 (NET-208-48-23-0-1)
                                  208.48.23.0 - 208.48.23.255
```

Network id#:5
```
Global Crossing Telecommunications, Inc.
   Olympia 6
   1213 NP Hilversum
   The Netherlands
```

Network id#:6
```
Global Crossing GBLX-5 (NET-208-178-0-0-1)
```

Network Data

```
                                  208.178.0.0 - 208.178.255.255
GC Internal Department FGC-REQ000000002985 (NET-208-178-174-0-1)
                                  208.178.174.0 - 208.178.174.255
```

Network id#:7

```
Global Crossing GBLX-11D (NET-64-212-0-0-1)
                                  64.212.0.0 - 64.215.255.255
Globalcrossing Internal GBX-REQ000000011413 (NET-64-214-65-0-1)
                                  64.214.65.0 - 64.214.65.255
```

Network id#:8

```
Global Crossing GBLX-6D (NET-208-50-192-0-1)
                                  208.50.192.0 - 208.51.255.255
GC Internal FGC-REQ000000005072 (NET-208-51-74-0-1)
                                  208.51.74.0 - 208.51.74.255
```

Network id#:9

```
OrgName:    Teleglobe Inc.
OrgID:      GLBE
Address:    1441 Carrie-Derick
City:       Montreal
StateProv:  QC
PostalCode: H3C-4S9
Country:    CA
```

Network id#:10

```
OrgName:    Teleglobe Inc.
OrgID:      GLBE
Address:    1441 Carrie-Derick
City:       Montreal
StateProv:  QC
PostalCode: H3C-4S9
Country:    CA
```

Network id#:11

```
OrgName:    Teleglobe Inc.
OrgID:      GLBE
Address:    1441 Carrie-Derick
City:       Montreal
StateProv:  QC
PostalCode: H3C-4S9
Country:    CA
```

Network id#:12

```
MIMOS BHD
   Technology Park Malaysia
   57000 Kuala Lumpur
```

Network id#:13

```
OrgName:    MIMOS
OrgID:      MIMOS
Address:    MIMOS Berhad, Technology Park Malaysia
Address:    Kuala Lumpur, 57000
City:
StateProv:
PostalCode:
Country:    MY
```

Network Data

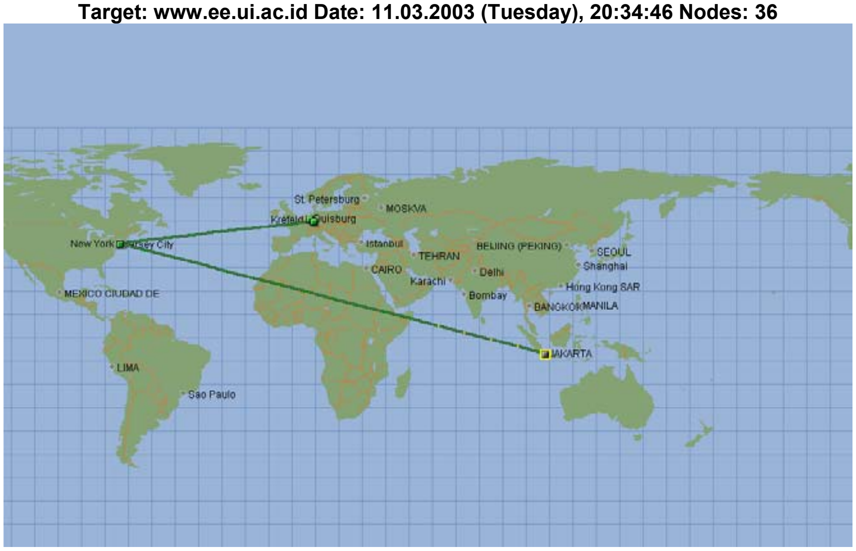
Network id#:14

```
Pusat Komputer
   Universiti Kebangsaan Malaysia
   43600 Bangi
   SELANGOR
```

APPENDIX – F

Internet Traffic Trace: Germany - Indonesia

Target: www.ee.ui.ac.id Date: 11.03.2003 (Tuesday), 20:34:46 Nodes: 36



Node Data

Node	Net	Reg	IP Address	Location	Node Name
1	-	-	192.168.192.3	Duisburg	clienta
2	1	-	192.168.192.1	-	
3	1	-	192.168.64.2	-	
4	1	-	192.168.80.2	-	
5	1	-	192.168.82.1	-	
6	2	1	134.91.90.1	-	cisbae1.uni-duisburg.de
7	2	1	134.91.3.3	-	routle-hrz.uni-duisburg.de
8	2	1	134.91.254.2	-	msfc65le.uni-duisburg.de
9	3	2	188.1.44.1	Krefeld	ar-essen1.g-win.dfn.de
10	3	2	188.1.86.1	Krefeld	cr-essen1.g-win.dfn.de
11	3	2	188.1.18.89	Wiesbaden	cr-frankfurt1.g-win.dfn.de
12	3	2	188.1.80.38	Wiesbaden	ir-frankfurt2.g-win.dfn.de
13	4	3	216.200.116.97	-	ge9-0.pr1.fra1.de.mfnx.net
14	4	3	216.200.116.213	-	so-0-1-0.cr1.fra1.de.mfnx.net
15	5	3	64.125.31.161	-	pos9-0.cr1.cdg2.fr.mfnx.net
16	6	3	212.69.164.1	-	so-4-0-0.cr1.lhr3.uk.mfnx.net
17	7	3	208.184.231.146	-	so-0-0-0.cr2.lhr3.uk.mfnx.net
18	5	3	64.125.31.182	-	so-7-0-0.cr2.lga1.us.mfnx.net
19	7	3	208.184.232.198	-	so-0-0-0.cr2.lga2.us.mfnx.net

Node Data					
20	5	4	64.124.232.2	Jersey City	core1-lga2-oc48.lga4.above.net
21	5	5	64.124.51.186	-	64.124.51.186.cogentco.com
22	8	5	66.28.4.177	-	g3-9.core02.jfk02.atlas.cogentco.com
23	8	5	66.28.4.86	-	p14-0.core02.ord01.atlas.cogentco.com
24	8	5	66.28.4.61	-	p15-0.core01.ord01.atlas.cogentco.com
25	8	5	66.28.4.42	-	p5-0.core01.sfo01.atlas.cogentco.com
26	8	5	66.28.4.94	-	p4-0.core01.sjc01.atlas.cogentco.com
27	8	5	66.28.65.198	-	g49.ba01.b006600-1.sjc01.atlas.cogentco.com
28	8	5	66.28.21.126	-	cyberstar.demarc.cogentco.com
29	9	-	202.161.130.42	-	
30	9	-	202.161.130.6	-	
31	9	-	202.161.128.194	-	
32	10	-	202.134.3.62	-	hssi6-0.396.ce-d2-cbi-gw1.hpbb.telkom.net.id
33	10	-	202.134.3.41	-	ge5-0.33.db-emm.hpbb.telkom.net.id
34	10	-	202.134.3.81	-	fe0-3-1.0.ix-sm2.hpbb.telkom.net.id
35	10	-	202.134.3.179	-	fe0-0-0.db-sm2.hpbb.telkom.net.id
36	11	-	152.118.101.8	JAKARTA	electron.ee.ui.ac.id

Network Data

Network id#:1

```
OrgName:    Internet Assigned Numbers Authority
OrgID:      IANA
Address:    4676 Admiralty Way, Suite 330
City:       Marina del Rey
StateProv:  CA
PostalCode: 90292-6695
Country:    US
```

Network id#:2

```
OrgName:    University of Duisburg
OrgID:      UNIVER-283
Address:    Lotharstrasse 65
City:       Duisburg, D-47048
StateProv:
PostalCode:
Country:    DE
```

Network id#:3

```
OrgName:    RIPE Network Coordination Centre
OrgID:      RIPE
Address:    Singel 258
Address:    1016 AB
City:       Amsterdam
StateProv:
PostalCode:
Country:    NL
```

Network id#:4

```
OrgName:    Abovenet Communications, Inc
OrgID:      ABVE
Address:    360 Hamilton Avenue
City:       White Plains
StateProv:  NY
PostalCode: 10601
Country:    US
```

Network Data

Network id#:5
```
AboveNet Deutschland GmbH
   Eschborner Landstrasse 112
   D-60489 Frankfurt
   Germany
```

Network id#:6
```
INCA Inh: Ing. Wolfgang Handl
   Fleischmanngasse 4/1a
   A-1070 Wien
   Austria
```

Network id#:7
```
OrgName:    Abovenet Communications, Inc
OrgID:      ABVE
Address:    360 Hamilton Avenue
City:       White Plains
StateProv:  NY
PostalCode: 10601
Country:    US
```

Network id#:8
```
OrgName:    Cogent Communications
OrgID:      COGC
Address:    1015 31st Street, NW
City:       Washington
StateProv:  DC
PostalCode: 20007
Country:    US
```

Network id#:9
```
2440 Research Blvd
   Rockville, MD 20850
```

Network id#:10
```
PT TELEKOMUNIKASI INDONESIA (DIVMEDIA)
   Jln Kebon Sirih No. 37
   JAKARTA 10340
```
Network id#:11
```
OrgName:    University of Indonesia
OrgID:      UNIVER-208
Address:    Computer Science Center
Address:    Kampus UI, Depok 16424
City:
StateProv:
PostalCode:
Country:    ID
```

APPENDIX - G

Conceptual Fans Configuration

```
#!/bin/sh
# FANS Virtual Network Testing Setup (f.setup)
# Configuration includes 3 tier topology
# Using: One pass on pipes.
#                   (MY)
#                 //    \\
#               /2        3\
#             1/            \4
#           //                \\
#         // ________5________  \\
#        (GE)--------6--------(INA)
#
# Configuration is for fullduplex connection to emulate WAN
# Channel parameter configuration

BW_GM="2Mbit/s"; # Bandwidth for German - Malaysia Channel
BW_GI="100Kbit/s"; # Bandwidth for German - Indonesia Channel
BW_MI="2Mbit/s"; # Bandwidth for Malaysia - Indonesia Channel
Delay_GM="175ms"; # Delay for German - Malaysia Channel
Delay_GI="0ms"; # Delay for German - Indonesia Channel
Delay_MI="270ms"; # Delay for Malaysia - Indonesia Channel
Q_GM="100"; # Queue for German - Malaysia Channel
Q_GI="100"; # Queue for German - Indonesia Channel
Q_MI="100"; # Queue for Malaysia - Indonesia Channel
BC_GM="64"; # Bucket Size for German - Malaysia Channel (Def 64, Max 1024)
BC_GI="64"; # Bucket Size for German - Indonesia Channel (Def 64, Max 1024)
BC_MI="64"; # Bucket Size for Malaysia - Indonesia Channel (Def 64, Max 1024)


#-------------------Starting Here, no modifications necessary-----
# Preparation and Network Address Assignment

fwcmd="/sbin/ipfw"
german="192.168.64.0/24"
german1="134.91.100.125"
indonesia="192.168.72.0/24"
malaysia="192.168.84.0/24"
othernet1="192.168.82.0/24"
othernet2="192.168.80.0/24"

# Flush out the list before begin.
/sbin/sysctl net.inet.ip.fw.one_pass=1
${fwcmd} -f flush

# Setting up topology and configure channel
${fwcmd} add pipe 1 all from ${german}    to ${malaysia} out
${fwcmd}     pipe 1 config bw ${BW_GM} delay ${Delay_GM} queue ${Q_GM} bucket ${BC_GM}

${fwcmd} add pipe 2 all from ${malaysia}  to ${german} out
${fwcmd}     pipe 2 config bw ${BW_GM} delay ${Delay_GM} queue ${Q_GM}  bucket ${BC_GM}

${fwcmd} add pipe 7 all from ${german1}    to ${malaysia} out
${fwcmd}     pipe 7 config bw ${BW_GM} delay ${Delay_GM} queue ${Q_GM} bucket ${BC_GM}

${fwcmd} add pipe 8 all from ${malaysia}  to ${german1}  out
${fwcmd}     pipe 8 config bw ${BW_GM} delay ${Delay_GM} queue ${Q_GM} bucket ${BC_GM}

${fwcmd} add pipe 3 all from ${malaysia}  to ${indonesia} out
${fwcmd}     pipe 3 config bw ${BW_MI} delay ${Delay_MI} queue ${Q_MI} bucket ${BC_MI}

${fwcmd} add pipe 4 all from ${indonesia} to ${malaysia} out
${fwcmd}     pipe 4 config bw ${BW_MI} delay ${Delay_MI} queue ${Q_MI} bucket ${BC_MI}
```

```
${fwcmd} add pipe 5 all from ${german}    to ${indonesia} out
${fwcmd}     pipe 5 config bw ${BW_GI} delay ${Delay_GI} queue ${Q_GI} bucket ${BC_GI}

${fwcmd} add pipe 6 all from ${indonesia} to ${german} out
${fwcmd}     pipe 6 config bw ${BW_GI} delay ${Delay_GI} queue ${Q_GI} bucket ${BC_GI}

${fwcmd} add pipe 9 all from ${german1}    to ${indonesia} out
${fwcmd}     pipe 9 config bw ${BW_GI} delay ${Delay_GI} queue ${Q_GI} bucket ${BC_GI}

${fwcmd} add pipe 10 all from ${indonesia} to ${german1} out
${fwcmd}     pipe 10 config bw ${BW_GI} delay ${Delay_GI} queue ${Q_GI} bucket ${BC_GI}

# Handling the rest of the network
${fwcmd} add allow ip from any to any
```

APPENDIX - H

Conceptual Fans Configuration

```
#!/bin/sh
# FANS Virtual Network Testing Setup (fa.setup)
# Configuration includes 3 tier topology
# Using Separated links
#                  (H)
#                   |
#                  (MY)
#                //    \\
#              /2        3\
#            1/            \4
#          //                \\
#        // ________5________  \\
# (H)--(GE) --------6-------- (INA)---(H)
#
# Configuration is for fullduplex connection to emulate WAN

# Channel parameter configuration
BW_G="10Mbit/s"; # Bandwidth for German - Malaysia Channel
BW_I="1Mbit/s"; # Bandwidth for German - Indonesia Channel
BW_M="2Mbit/s"; # Bandwidth for Malaysia - Indonesia Channel
Delay_GM="175ms"; # Delay for German - Malaysia Channel
Delay_GI="0ms"; # Delay for German - Indonesia Channel
Delay_MI="270ms"; # Delay for Malaysia - Indonesia Channel
Q_GM="100"; # Queue for German - Malaysia Channel
Q_GI="100"; # Queue for German - Indonesia Channel
Q_MI="100"; # Queue for Malaysia - Indonesia Channel
Q_G="100"; # Queue for German
Q_M="100"; # Queue for Malaysia
Q_I="100"; # Queue for Indonesia
BC_GM="64"; # Bucket Size for German - Malaysia Channel (Def 64, Max 1024)
BC_GI="64"; # Bucket Size for German - Indonesia Channel (Def 64, Max 1024)
BC_MI="64"; # Bucket Size for Malaysia - Indonesia Channel (Def 64, Max 1024)


#-------------------Starting Here, no modifications necessary-----
# Preparation and Network Address Assignment

fwcmd="/sbin/ipfw"
german="192.168.64.0/24"
indonesia="192.168.72.0/24"
malaysia="192.168.84.0/24"
othernet1="192.168.82.0/24"
othernet2="192.168.80.0/24"

# Flush out the list before begin.
/sbin/sysctl net.inet.ip.fw.one_pass=0
${fwcmd} -f flush

# Setting up topology and configure channel

${fwcmd} add pipe 1 all from ${german}  to any in
${fwcmd}     pipe 1 config bw ${BW_G} queue ${Q_G}
${fwcmd} add pipe 2 all from ${malaysia} to any in
${fwcmd}     pipe 2 config bw ${BW_M} queue ${Q_M}
${fwcmd} add pipe 3 all from ${indonesia} to any in
${fwcmd}     pipe 3 config bw ${BW_I} queue ${Q_I}

${fwcmd} add pipe 4 all from ${german}    to ${malaysia} out
${fwcmd}     pipe 4 config delay ${Delay_GM} queue ${Q_GM} bucket ${BC_GM}
${fwcmd} add pipe 5 all from ${malaysia}  to ${german} out
${fwcmd}     pipe 5 config delay ${Delay_GM} queue ${Q_GM}  bucket ${BC_GM}
```

```
${fwcmd} add pipe 6 all from ${malaysia}  to ${indonesia} out
${fwcmd}     pipe 6 config delay ${Delay_MI} queue ${Q_MI} bucket ${BC_MI}
${fwcmd} add pipe 7 all from ${indonesia} to ${malaysia} out
${fwcmd}     pipe 7 config delay ${Delay_MI} queue ${Q_MI} bucket ${BC_MI}
${fwcmd} add pipe 8 all from ${german}    to ${indonesia} out
${fwcmd}     pipe 8 config delay ${Delay_GI} queue ${Q_GI} bucket ${BC_GI}
${fwcmd} add pipe 9 all from ${indonesia} to ${german} out
${fwcmd}     pipe 9 config delay ${Delay_GI} queue ${Q_GI} bucket ${BC_GI}

${fwcmd} add pipe 10 all from any   to ${german} out
${fwcmd}     pipe 10 config bw ${BW_G} queue ${Q_G}
${fwcmd} add pipe 11 all from any to ${malaysia} out
${fwcmd}     pipe 11 config bw ${BW_M} queue ${Q_M}
${fwcmd} add pipe 12 all from any    to ${indonesia} out
${fwcmd}     pipe 12 config bw ${BW_I} queue ${Q_I}

# Handling the rest of the network
${fwcmd} add allow ip from any to any
```

APPENDIX - I

Conceptual Fans Configuration

```
#!/bin/sh
# FANS Virtual Network Testing Setup (fan.setup)
# Configuration includes 3 tier topology
# Using: Separated links and packet reordering
#                   (H)
#                    |
#                   (MY)
#                 //    \\
#               /2        3\
#             1/            \4
#           //                \\
#         // ________5________  \\
# (H)--(GE) --------6-------- (INA)---(H)
#
# Configuration is for fullduplex connection to emulate WAN
# Channel parameter configuration

BW_G="2Mbit/s"; # Bandwidth for German - Malaysia Channel
BW_I="400Kbit/s"; # Bandwidth for German - Indonesia Channel
BW_M="2Mbit/s"; # Bandwidth for Malaysia - Indonesia Channel

# Delay for German - Malaysia Channel
Delay_GM="175ms"; Delay_GM0="160ms"; Delay_GM1="190ms"
# Delay for German - Indonesia Channel
Delay_GI="360ms"; Delay_GI0="330ms"; Delay_GI1="390ms"
# Delay for Malaysia - Indonesia Channel
Delay_MI="270ms"; Delay_MI0="250ms"; Delay_MI1="290ms"

Q_GM="100"; # Queue for German - Malaysia Channel
Q_GI="100"; # Queue for German - Indonesia Channel
Q_MI="100"; # Queue for Malaysia - Indonesia Channel
BC_GM="64"; # Bucket Size for German - Malaysia Channel (Def 64, Max 1024)
BC_GI="64"; # Bucket Size for German - Indonesia Channel (Def 64, Max 1024)
BC_MI="64"; # Bucket Size for Malaysia - Indonesia Channel (Def 64, Max 1024)

#-------------------Starting Here, no modifications necessary-----
# Preparation and Network Address Assignment

fwcmd="/sbin/ipfw"
german="192.168.64.0/24"
german1="134.91.100.164"
indonesia="192.168.72.0/24"
malaysia="192.168.84.0/24"
othernet1="192.168.82.0/24"
othernet2="192.168.80.0/24"

# Flush out the list before begin.
/sbin/sysctl net.inet.ip.fw.one_pass=0
${fwcmd} -f flush

# Setting up topology and configure channel
# Configuring Outgoing channel of endpoint

${fwcmd} add 100 pipe 1 all from ${german}  to any in
${fwcmd}     pipe 1 config bw ${BW_G}
${fwcmd} add 101 pipe 2 all from ${malaysia} to any in
${fwcmd}     pipe 2 config bw ${BW_M}
${fwcmd} add 102 pipe 3 all from ${indonesia} to any in
${fwcmd}     pipe 3 config bw ${BW_I}

# Multipath on German Malaysia Link
${fwcmd} add 201 prob 0.33 skipto 210 all from ${german}    to ${malaysia} out
```

```
${fwcmd} add 202 prob 0.5 skipto 220 all from ${german}    to ${malaysia} out
${fwcmd} add 203 skipto 230 all from ${german}    to ${malaysia} out
${fwcmd} add 204 skipto 240 all from any   to any

${fwcmd} add 210 pipe 21 all from ${german}    to ${malaysia} out
${fwcmd}         pipe 21 config delay ${Delay_GM} queue ${Q_GM} bucket ${BC_GM}
${fwcmd} add 211 skipto 600 all from ${german}    to ${malaysia} out

${fwcmd} add 220 pipe 22 all from ${german}    to ${malaysia} out
${fwcmd}         pipe 22 config delay ${Delay_GM0} queue ${Q_GM} bucket ${BC_GM}
${fwcmd} add 221 skipto 600 all from ${german}    to ${malaysia} out

${fwcmd} add 230 pipe 23 all from ${german}    to ${malaysia} out
${fwcmd}         pipe 23 config delay ${Delay_GM1} queue ${Q_GM} bucket ${BC_GM}
${fwcmd} add 231 skipto 600 all from ${german}    to ${malaysia} out

${fwcmd} add 241 prob 0.33 skipto 250 all from ${malaysia}    to ${german} out
${fwcmd} add 242 prob 0.5 skipto 260 all from ${malaysia}    to ${german} out
${fwcmd} add 243 skipto 270 all from ${malaysia}    to ${german} out
${fwcmd} add 244 skipto 280 all from any   to any

${fwcmd} add 250 pipe 24 all from ${malaysia}  to ${german} out
${fwcmd}         pipe 24 config delay ${Delay_GM} queue ${Q_GM}  bucket ${BC_GM}
${fwcmd} add 251 skipto 600 all from ${malaysia}    to ${german} out

${fwcmd} add 260 pipe 25 all from ${malaysia}  to ${german} out
${fwcmd}         pipe 25 config delay ${Delay_GM0} queue ${Q_GM}  bucket ${BC_GM}
${fwcmd} add 261 skipto 600 all from ${malaysia}    to ${german} out


${fwcmd} add 270 pipe 26 all from ${malaysia}  to ${german} out
${fwcmd}         pipe 26 config delay ${Delay_GM1} queue ${Q_GM}  bucket ${BC_GM}
${fwcmd} add 271 skipto 600 all from ${malaysia}    to ${german} out

# Multipath on Indonesia Malaysia Link
${fwcmd} add 301 prob 0.33 skipto 310 all from ${malaysia}    to ${indonesia} out
${fwcmd} add 302 prob 0.5 skipto 320 all from ${malaysia}    to ${indonesia} out
${fwcmd} add 303 skipto 330 all from ${malaysia}    to ${indonesia} out
${fwcmd} add 304 skipto 340 all from any   to any

${fwcmd} add 310 pipe 31 all from ${malaysia}    to ${indonesia} out
${fwcmd}         pipe 31 config delay ${Delay_MI} queue ${Q_MI} bucket ${BC_MI}
${fwcmd} add 311 skipto 600 all from ${malaysia}    to ${indonesia} out

${fwcmd} add 320 pipe 32 all from ${malaysia}    to ${indonesia} out
${fwcmd}         pipe 32 config delay ${Delay_MI0} queue ${Q_MI} bucket ${BC_MI}
${fwcmd} add 321 skipto 600 all from ${malaysia}    to ${indonesia} out

${fwcmd} add 330 pipe 33 all from ${malaysia}    to ${indonesia} out
${fwcmd}         pipe 33 config delay ${Delay_MI1} queue ${Q_MI} bucket ${BC_MI}
${fwcmd} add 331 skipto 600 all from ${malaysia}    to ${indonesia} out

${fwcmd} add 341 prob 0.33 skipto 350 all from ${indonesia}    to ${malaysia} out
${fwcmd} add 342 prob 0.5 skipto 360 all from ${indonesia}    to ${malaysia} out
${fwcmd} add 343 skipto 370 all from ${indonesia}    to ${malaysia} out
${fwcmd} add 344 skipto 380 all from any   to any

${fwcmd} add 350 pipe 34 all from ${indonesia}  to ${malaysia} out
${fwcmd}         pipe 34 config delay ${Delay_MI} queue ${Q_MI}  bucket ${BC_MI}
${fwcmd} add 351 skipto 600 all from ${indonesia}    to ${malaysia} out

${fwcmd} add 360 pipe 35 all from ${indonesia}  to ${malaysia} out
${fwcmd}         pipe 35 config delay ${Delay_MI0} queue ${Q_MI}  bucket ${BC_MI}
${fwcmd} add 361 skipto 600 all from ${indonesia}    to ${malaysia} out

${fwcmd} add 370 pipe 36 all from ${indonesia}  to ${malaysia} out
${fwcmd}         pipe 36 config delay ${Delay_MI1} queue ${Q_MI}  bucket ${BC_MI}
${fwcmd} add 371 skipto 600 all from ${indonesia}    to ${malaysia} out

# Multipath on German Indonesia Link
```

```
${fwcmd} add 401 prob 0.33 skipto 410 all from ${german}    to ${indonesia} out
${fwcmd} add 402 prob 0.5 skipto 420 all from ${german}    to ${indonesia} out
${fwcmd} add 403 skipto 430 all from ${german}    to ${indonesia} out
${fwcmd} add 404 skipto 440 all from any   to any

${fwcmd} add 410 pipe 41 all from ${german}    to ${indonesia} out
${fwcmd}         pipe 41 config delay ${Delay_GI} queue ${Q_GI} bucket ${BC_GI}
${fwcmd} add 411 skipto 600 all from ${german}    to ${indonesia} out

${fwcmd} add 420 pipe 42 all from ${german}    to ${indonesia} out
${fwcmd}         pipe 42 config delay ${Delay_GI0} queue ${Q_GI} bucket ${BC_GI}
${fwcmd} add 421 skipto 600 all from ${german}    to ${indonesia} out

${fwcmd} add 430 pipe 43 all from ${german}    to ${indonesia} out
${fwcmd}         pipe 43 config delay ${Delay_GI1} queue ${Q_GI} bucket ${BC_GI}
${fwcmd} add 431 skipto 600 all from ${german}    to ${indonesia} out

${fwcmd} add 441 prob 0.33 skipto 450 all from ${indonesia}    to ${german} out
${fwcmd} add 442 prob 0.5 skipto 460 all from ${indonesia}    to ${german} out
${fwcmd} add 443 skipto 470 all from ${indonesia}    to ${malaysia} out
${fwcmd} add 444 skipto 480 all from any   to any


${fwcmd} add 450 pipe 44 all from ${indonesia}  to ${german} out
${fwcmd}         pipe 44 config delay ${Delay_GI} queue ${Q_GI}  bucket ${BC_GI}
${fwcmd} add 451 skipto 600 all from ${indonesia}    to ${german} out

${fwcmd} add 460 pipe 45 all from ${indonesia}  to ${german} out
${fwcmd}         pipe 45 config delay ${Delay_GI0} queue ${Q_GI}  bucket ${BC_GI}
${fwcmd} add 461 skipto 600 all from ${indonesia}    to ${german} out

${fwcmd} add 470 pipe 46 all from ${indonesia}  to ${german} out
${fwcmd}         pipe 46 config delay ${Delay_GI1} queue ${Q_GI}  bucket ${BC_GI}
${fwcmd} add 471 skipto 600 all from ${indonesia}    to ${german} out

# Configuring incoming channel to end point

${fwcmd} add 700 pipe 10 all from any   to ${german} out
${fwcmd}     pipe 10 config bw ${BW_G}

${fwcmd} add 701 pipe 11 all from any to ${malaysia} out
${fwcmd}     pipe 11 config bw ${BW_M}

${fwcmd} add 702 pipe 12 all from any    to ${indonesia} out
${fwcmd}     pipe 12 config bw ${BW_I}

# Handling the rest of the network
${fwcmd} add allow ip from any to any
```

APPENDIX - J

Reviewer's Comments on Papers

Ramli, K., Ekadiyanto, F.A, Hunger, A., "Utilizing ALAN Concept to Improve the Performance of Streaming Multimedia Applications over Heterogeneous Bandwidth Environment", to appear in *Springer's Lecture Notes on Computer Science: Proceedings of Worskshop on Active Network Technology and Applications*, Osaka, Japan, May, 2003, pp 129-140.

Reviewer's Comments:

Paper No: ANTA2003-12

Title: Utilizing ALAN Concept To Improve The Performance of Streaming Multimedia Applications over Heterogeneous Bandwidth Environment

Author(s): Kalamullah Tamli, Fransiskus A Ekadiyanto, Axel Hunger

Evaluation of work and contribution:
08 = Good solid work of some importance

Significance to theory and practice:
08 = Highly significant

Originality novelty:
07 = A pioneering piece of work

Relevance to the call of papers:
08 = Definitely relevant

Readability and organization:
08 = Basically well written

Overall recommendation:
08 = Accept (good quality)

Reviewer familiarity:
07 = Good knowledge

Comments to author:

Summary:

Paper No: 12
Title: Utilizing ALAN Concept to improve the performance of streaming multimedia application over the heterogeneous bandwidth environment
Author(s): Kallamulah Ramli et al.

Evaluation of work and contribution:
10/09 = Excellent work and a major contribution
08/07 = Good solid work of some importance ----------X
06/05 = Solid work but marginal contribution
04/03 = Marginal work but minor contribution
02/01 = Very questionable work and contribution

Significance to theory and practice:
10/09 = Very high significance
08/07 = Highly significant ---------X
06/05 = Not bad
04/03 = Only little significance
02/01 = Absolutely no relevance

Originality novelty:
10/09 = Trailblazing
08/07 = A pioneering piece of work ---------------X
06/05 = One step ahead of the pack
04/03 = YAPA (yet another paper about...)
02/01 = Its been said many times before

Relevance to the call of papers:
10/09 = Right on target
08/07 = Definitely relevant-------------X
06/05 = Close enough
04/03 = Not really appropriate
02/01 = Definitely inappropriate

Readability and organization:
10/09 = Very good ------------------X
08/07 = Basically well written
06/05 = Readable
04/03 = Needs considerable work
02/01 = Unacceptably bad

Overall recommendation:
10/09 = Definitely accept (very high quality)

08/07 = Accept (good quality) ----------------------X

06/05 = Accept if room (marginal quality)

04/03 = Likely reject (low quality)

02/01 = Definitely reject (has no merit)

Reviewer familiarity:

10/09 = Very familiar with the subject

08/07 = Good knowledge

06/05 = Moderately familiar------------X

04/03 = Marginally acquainted

02/01 = Not acquainted

Comments to author:

This paper shows FANS(user Friendly Active Network) capabilties by experiments. Please write an introduction of FANS more precisely in the full paper, so that the readers understand easily how the FANS capabilities lead to this conclusion.

Ramli, K., Hunger, A., Erdani, Y., "A User Interface for Dynamic Multiple Locations Streaming Multimedia Environment", *Proceedings of IASTED Computer Science and Technology Conference*, Cancun, Mexico, May, 2003, pp. 348-352.

Reviewer's Comments
Conference Title: Computer Science and Technology (CST 2003)

Paper Number: 394-081
Paper Title: A User Interface for Dynamic Multiple Location Streaming Multimedia Environment

March 10, 2003

Kalamullah Ramli
Universitaet Duisburg-Esen, Duisburg NRW
Germany 47057

Re: 394-081
A User Interface for Dynamic Multiple Location Streaming Multimedia Environment

All conference registration material is available at the following web site address: http://www.iasted.org/conferences/2003/cancun/cst.htm

Dear Mr. Ramli,

Congratulations! Your paper has been accepted for presentation at the IASTED International Conference on Computer Science and Technology (CST 2003), which will be held May 19 to May 21, 2003, in Cancun, Mexico. We cordially invite you to attend the conference and to present your paper.

The numerical list below is a list of the reviewers' averaged score for the criteria of originality, contribution, references, presentation and language for your respected paper. Please note that the final result of the paper is in no way solely dependent upon the numerical scores assigned by the reviewers. Other factors such as the total number of submissions, comments and relevance to conference topic area also influence the final result. Also, please note that some reviewers may choose not to state any comments.

(Key: 1=Excellent, 2=Good, 3=Satisfactory, 4=Unsatisfactory, 5=Poor)
Originality: 3
Contribution: 3
References: 2.5
Presentation: 2.5
Language: 3.5

Final Result: Accepted

Comments:
Contribution:
This paper develops a user interface that is suitable for dynamic and multiple locations streaming multimedia environment. The user interface conforms the five attributes identified for such environment.

The result and discussion section is very short. The authors needs to do more experiments to evaluate their proposed solution.

The paper is well written and clear, and is good for a conference submission.

Suggestions:
The results and discussion section is very short. The authors needs to do more experiments to evaluate their proposed solution.

Contribution:
This paper presents the design of user interface for multimedia application. The architecture and design tips are discussed. Several examples are shown to explain the results.

Suggestions:
The structure of the whole system is not presented clearly. The correlation of the system functions and user-interface design is not stated. The unique property of the system is not clear. How to solve the problems that might be encountered in real application is not mentioned. The discussion part is weak.

APPENDIX - K

JMF Engineering People Comments on Synchronization Issues

1. Regarding QoS Definition and Parameter

David Rivas (Java Media Engineering)
http://swjscmail1.java.sun.com/cgi-bin/wa?A2=ind9702&L=jmf-interest&P=R1086

- What are you doing about QOS?

We spent some time evaluating the various ways of supporting QOS specification from an API users standpoint.

We decided that there was not enough agreement among the various communities interested in Multimedia QOS about what information a user should provide to describe QOS desires or requirements. Rather than specify an arbitrary measure we decided to wait for this technology to mature.

Players for protocols that are capable of affecting and controlling QOS can be developed and the interface for controlling QOS on a Player could be exposed. As consensus on what such an API might look like develops we can consider adding a specific Interface that Players could then choose to support.

- Are you going to help me build Players?

Yes. This effort was initially focused on API for using Players and as we build more Players we are converging on definitions for support Classes and Interfaces to make implementation easier.

- Its tacit assumption is that an application will always deal with "sources" that are well-named single flow endpoints (e.g a stored mpeg file referred to by its URL, etc.). In other words, the model is very client-centric, it assumes that the client has to deal with grouping of multiple flows coming from the same or different servers, it has to deal with their synchronization etc. In some cases, greater synchronization could be achieved if the source flows were grouped at the server end.

I would restate this as "protocol-centric" rather than "client-centric". A server that handles sync, and control and requires only simple client rendering will require a protocol to communicate the client's intents. A Player specific to that protocol would be built and communicate with a server to manage the media asset on the server. The client would only be responsible for rendering the data locally.

2. Regarding synchronization timing problem (TimeBase, etc.)

David Rivas (Java Media Engineering, JavaSoft)
http://swjscmail1.java.sun.com/cgi-bin/wa?A2=ind9704&L=jmf-interest&P=R413

There is no scheduler in the JMF beyond the thread scheduling offered by the VM. Since the VM does not provide real-time services such as deadline-driven scheduling or task admission control, the kinds of guarantees that you are asking for are not yet possible.

The JMF provides mechanisms for expressing timing relationships between media streams, and devices or software that process media. The synchronization provided by a Clock is dependent on the particular clock implementation, the mix of threads in the VM at the particular time synchronization operations occur, the underlying operating system that the VM is running on, and the speed and type of processor that that VM is running on. This is exactly what you would expect to effect the performance of a real-time process in a non real-time operating environment.

Currently, the JMF provides no more support for scheduling execution of arbitrary code than what is supported in the existing VM and Java environment. We are considering providing scheduling mechanisms that are based on the passage of time in a Clock. If this kind of service is provided, no greater guarantee on the accuracy of the invocation can be provided than is currently provided throughout the rest of platform. To provide for any hard definition of scheduling at "exact intervals" would require more support from the VM than existing implementations provide.

----

http://swjscmail1.java.sun.com/cgi-bin/wa?A2=ind9908&L=jmf-interest&P=R5719&m=3509
Hi Zubair,
synchronization between media streams is accomplished in the multiplexer (server side), i.e. it is handled internally. In case of a media source with video and audio tracks the audio track would drive the transmission of data. There are many possibilities for the client side not receiving data or experiencing delays etc. Change your system parameters and observe the results. If you don't see any improvements please send us a detailed description of your system (platform, OS, network, media type etc.) and we will check it out.
Regards,

---->Marc <Marc.Owerfeldt@ENG.SUN.COM>

----

http://swjscmail1.java.sun.com/cgi-bin/wa?A2=ind9908&L=jmf-interest&P=R5776&m=3509
Lack of synchronization at the receiver can also result from the audio and video streams not being synchronized at the source(s). For example, if two separate programs are used to create the audio and video streams and those programs don't coordinate with each other to reference a common time base for the timestamps they send in RTCP SR packets, or if the timestamps are not generated properly, then the receiver can't synchronize the streams.

-- Steve < Stephen Casner casner@CISCO.COM >

----

http://swjscmail1.java.sun.com/cgi-bin/wa?A2=ind9908&L=jmf-interest&P=R5814&m=3509
Hi Zubair,

the current release of JMF doensn't support frame dropping in the JPEG Encoder. This may cause the synchronization problem you described. We will enable frame dropping in the upcoming JMF 2.0 Beta release (in few weeks).

Of course Stephen's comment is also correct. If you don't have synchronized data sources (i.e. using vic and vat on the server) the result on the client side will be out of synch, too.
Regards,

----> Marc <Marc.Owerfeldt@ENG.SUN.COM>

----

http://swjscmail1.java.sun.com/cgi-bin/wa?A2=ind9908&L=jmf-interest&P=R18413&m=3509

Ivan Wong had suggested the following:
> The timestamps on different captured streams are supposed to be stamped
> with the actual captured time w.r.t. the same time base: SystemTimeBase.
> That can then be use for synchronization. But for now, the timestamps
> are stamped with TIME_UNKNOWN.

is this still true with 2.0 beta? if so, when can we expect this? thanks.

Reha

Sorry Reha. We know this is a crucial feature for a lot of our customers. But we didn't manage to fix that in 2.0beta because of other complexities involved.

We should be able to fix that in the next release (hopefully not as long as the EA-Beta time gap).

Ivan Wong <ivg@ROM12.ENG.SUN.COM>
Java Media Engineering
Sun Microsystems, Inc.

Other comments to this problem

http://swjscmail1.java.sun.com/cgi-bin/wa?A2=ind9704&L=jmf-interest&P=R544

>> We are considering providing scheduling mechanisms that are based on the
>> passage of time in a Clock. If this kind of service is provided, no \
>> greater guarantee on the accuracy of the invocation can be provided than
>> is currently provided throughout the rest of platform...
>
>Even without an iron-clad guarantee of accuracy, having the notion of a
>real-time clock would be a big boon to Java. I hope you do decide to
>provide this type of scheduling.

Totally agree. This is what the real-time-java@iastate.edu folks have been saying for a while. I would assume that its a platform issue (or the VM needs a significant mod).

Btw, kudos to the JMF folks at JavaOne. Very impressive work so far. Keep it up! :)

Frank G.
 - licensing www.headspace.com's stuff was a great move also...

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+=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=+
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```