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# "Adaptive Routing for Communication Network using Client System"

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## Abstract

Problems with bearing multimedia flows on IP networks are mainly related to the bandwidth they require and to the strict maximum delay requirements that must be met. This point is particularly important when multimedia applications have to provide users with real-time interaction. As the paradigm for computing moves from centralized compute-servers to

distributed systems, the experience base for bulk-data and a transactional protocol is expanding. Current networks enable large quantities of data to be shared using application programs that make the underlying data delivery service transparent to the users. Windowing software environments for workstations decouple servers from displays, using many short messages and fixed-size page transfers to coordinate a session between the server and the display. Yet there are certain applications that produce data in a continuous stream for which neither bulk data transfer nor a transactional service is appropriate. Multimedia describes the emerging network service that unifies into one provider the ability to serve reliable data exchanges and transactional interactions, as well as audio and video transfers. The term digital continuous media communication refers to data transfer where the source produces an arbitrarily long stream of data at a fixed rate such that the data is intended for playback at the receiving end. It is this aspect of playback that prevents continuous-media data transfers from using conventional network services.

## I- INTRODUCTION

Networking infrastructure nowadays encourages integrating the multimedia services offered over the different networking platforms. Multimedia streaming facilitates efficient interactivity and retrieval of media sources across different platforms and distributed systems. Globally speaking, a multimedia communication system can be classified into a body and its peripherals, or, in other words, a core and edges. The core of the multimedia communication systems refers to the networking infrastructure, the link capacities and the routing protocols. The edges are characterized by the multimedia traffic sources, the multi-participant communication

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environment and the user interfaces. Nowadays, most systems include multiple clock domains. Several different needs are driving the research in novel clocking and synchronization methods for complex systems-on-chip (SoCs).

In terms of engineering effort, silicon and power, it is expensive to maintain the globally synchronous assumption since the number of clock tree leaves roughly doubles with every new technology node. The buy and assemble model of building SoCs is making systems increasingly modular. Hierarchical physical design is required to avoid costly timing closure iterations for every small change in any part of the design; such iterations drive up the NRE costs of SoCs and impede and discourage design space exploration.

Hierarchical physical design can be guaranteed with a truly latency-insensitive design style. Variations due to manufacturing and operating conditions require adaptive synchronization techniques. Power management techniques like per-module dynamic voltage frequency scaling (DVFS) are necessary to guarantee low-power operation. Their deployment demands the ability to safely cross clock domain boundaries between clocks running at different clock frequencies. The interfaces must have limited overheads and must be easily integrated in the standard EDA design flow.

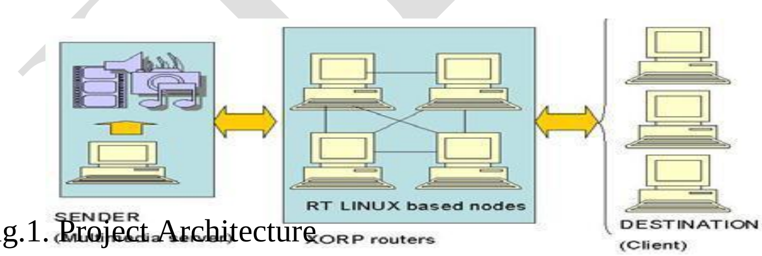


Fig.1. Project Architecture

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The interfaces must guarantee maximal throughput and low latency, to ensure high performances. This is particularly important for systems in which latency determines throughput, such as Networks-on-Chip. A network environment includes commonly operated information and communication networks which are part of a telecommunication network.

It constitutes a framework for a platform generally used for provision of various multimedia services. Increase in computing power of processors implemented in network stations stimulates the increase in a set of different points of access to generally accessible wide area network resources. At the same time, the network environment opens for evolutionary changes in the process of telecommunication services performance.

Today, the representative of the modern society uses in its everyday life more and more technical "innovations" such as technology advanced phones, pockets, smart phones and tablets. As part of the network operator service package, they offer numerous services. Except for the possibility of access to web sites, e-mail and transfer of files, the telephone services covering image and sound transfer gain increasingly more interest. Thus, there is an economically justified need to evaluate the functionality of the environment that provides media services that are the result of numerous network properties.

## II- FUNCTIONALITY

The functionality of services may be presented from different perspectives. One of them is focused on practical use and depends on such components as availability, accessibility and

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continuity. Additionally, the functionality of services may be affected by a number of other properties such as intuitive use, quick operation etc.

Considering the conclusions drawn from both the experience related to operation of real network environments and the analysis of published scientific works as well as standardization documents, for the purposes of functionality assessment, it is necessary to limit the set of factors considered to only the most important, i.e. to delays in delivering packets, data flow capacity, delays fluctuations, packet loss, packet duplication, change of delivered packets sequence, error rate.

Furthermore, elements that the user has direct contact with are also considered, such as the number of essential service applications and the number of network access points. The condition for achieving and ensuring functionality of services is the stability and correctness of the hardware and software platform containing network components of data transfer (i.e. telecommunications switches, backbone routers, wire and wireless links etc.) and providing services (i.e. servers). An important determinant is the proper configuration of devices as well as system and application software. Various applications have different network requirements, e.g.:

1.Audio and video transmission requires small delays but tolerates partial packet loss.

2.Transfer of data does not require continuity of transmission, however all packets must be delivered using a large bandwidth.

3.In the case of instant messengers, a transmission without delays must be ensured and large bandwidth is not required.

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It can be noticed that the diversity of applications and devices used by network users forces the service providers to ensure time-variable network availability. The last element that may have an impact on services functionality is the technical aspect, i.e. the place and method of connection with network and network equipment (and its redundancy) used by the operator. Wire connection ensures sufficient bandwidth and stability, whereas wireless connections provide mobility.

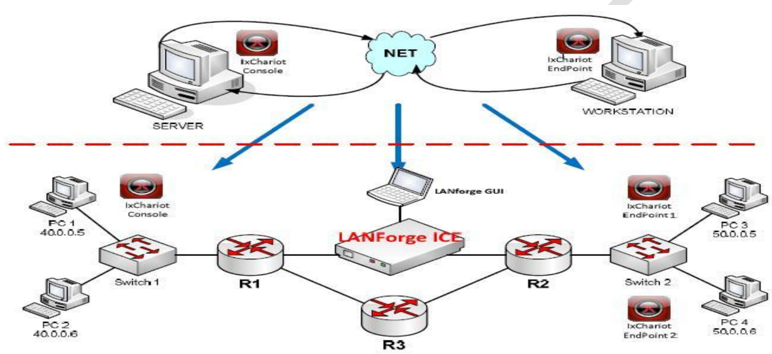


Fig.2. Generalized architecture of research

Environment

**LAN Network Technologies**

**Ethernet at 10 Mbps**

There is a vast installed base (about 40 million Ethernet nodes) of 10 Mbps Ethernet and 4 or 16 Mbps Token Ring Local Area Networks (LANs) using coaxial cable or twisted copper wire. Most new LANs used twisted copper wire cable. Ethernet uses a contention method to

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enable workstations attached to the same cable to share the data bandwidth on the cable. Nodes transmit to the network on demand, but continually monitor the network to see if another node is transmitting at the same time.

If this occurs both nodes cease transmission, and try again later at random intervals. The throughput with such a system is limited to about half the available bandwidth or 5 to 6 Mbps. Multimedia file servers can be attached to such LANs. Both Novell and Microsoft have developed software to support video on these traditional LANs. Video in an Intel DVI or MPEG form requires 1 to 2 Mbps per user, so it is clear that only a handful of users can run video applications simultaneously. Audio requires lower bandwidth but is sensitive to unpredictable delays.

For instance an intensive file transfer or video stream could prevent an audio application from transmitting onto the LAN for several tens of milliseconds, which is sufficient to reduce the intelligibility of voice. Improved performance is possible if LAN segments are divided up into segments with only a few attached users. Switched Ethernet takes this approach to the limit by enabling only one user to access a single segment which is then connected to a higher capacity network at a central hub by a switch.

The price per port of Ethernet switches ranges from £400 to £1000. The use of a switch permits the filtering of packets based on address. Switching times need to be fast enough to deliver packets at the basic 10 Mbps Ethernet line rate. An Ethernet switch would be equipped with a higher speed ATM or FDDI interface to other networks. However even with one user per

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Ethernet Switch port or segment contention between different applications on the same machine or incoming and outgoing traffic can occur.

**Fast Ethernet**

Most of the 10 Mbps parameters including the CSMA/CD media access protocol will remain unchanged and the standard is being named 100Base-T. Another 100 Mbps standard called 100VG-AnyLAN [LAN95] has also been proposed and changes the media access protocol to a demand priority system. Fast Ethernet products are only just starting to appear. Use of both standards will require a change out of workstation and hub cards. The cost of 100 Mbps Ethernet cards will be targeted to be comparable with high performance 10Base-T

**FDDI**

The Fiber Distributed Data Interface (FDDI) [Minoli93] was the only standards based technology operating at 100 Mbps for some time. FDDI has experienced slow market penetration due to the high cost of cards, still around £600. The first version of FDDI was developed as a campus trunk network for data. Ability to allocate bandwidth dynamically so that both 'synchronous and asynchronous' services can be provided. An upgraded FDDI standard called FDDI II has been designed. In addition to the data, packet switched mode in FDDI an isochronous circuit switched service is made available by imposing a 125 micro second frame structure. The 100 Mbps bandwidth can be split between packet data and up to fifteen isochronous channel operating at 6.144 Mbps each.

**MULTIMEDIA AND INTERNET PROTOCOLS**

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Multimedia refers to an electronically delivered combination of media including video, still images, audio, text in such a way that can be accessed interactively. Much of the content on the web today falls within this definition as understood by millions. Multimedia may be broadly divided into linear and non-linear categories. Linear active content progresses without any navigation control for the viewer such as a cinema presentation. Non-linear content offers user interactivity to control progress as used with a computer game or used in self-paced computer based training. Non-linear content is also known as hypermedia content.

Multimedia presentations can be live or recorded. A recorded presentation may allow interactivity via a navigation system. A live multimedia presentation may allow interactivity via interaction with the presenter or performer. Multimedia presentations may be viewed in person on stage, projected, transmitted, or played locally with a media player. A broadcast may be a live or recorded multimedia presentation.

Broadcasts and recordings can be either analog or digital electronic media technology. Digital online multimedia may be downloaded or streamed. Streaming multimedia may be live or on-demand. Multimedia games and simulations may be used in a physical environment with special effects, with multiple users in an online network, or locally with an offline computer, game system, or simulator.

Enhanced levels of interactivity are made possible by combining multiple forms of media content. But depending on what multimedia content you have it may vary. Online multimedia is increasingly becoming object-oriented and data-driven, enabling applications with collaborative end-user innovation and personalization on multiple forms of content over time. Examples of

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these range from multiple forms of content on web sites like photo galleries with both images (pictures) and title (text) user-updated, to simulations whose co-efficient, events, illustrations, animations or videos are modifiable, allowing the multimedia "experience" to be altered without reprogramming. Multimedia finds its application in various areas including, but not limited to, advertisements, art, education, entertainment, engineering, medicine, mathematics, business, scientific research and spatial, temporal applications.

A few application areas of multimedia are Creative industries, Commercial, Entertainment and Fine Arts, Education, Engineering, Industry, Mathematical and Scientific Research, Medicine and Multimedia in Public Places.

**Internet Control Message Protocol (ICMP)**

ICMP is network diagnostic and error reporting protocol. ICMP belongs to IP protocol suite and uses IP as carrier protocol. After constructing ICMP packet, it is encapsulated in IP packet. Because IP itself is a best-effort non-reliable protocol, so is ICMP. Any feedback about network is sent back to the originating host. If some error in the network occurs, it is reported by means of ICMP. ICMP contains dozens of diagnostic and error reporting messages. ICMP-echo and ICMP-echo-reply are the most commonly used ICMP messages to check the reachability of end-to-end hosts. When a host receives an ICMP-echo request, it is bound to send back an ICMP-echo-reply. If there is any problem in the transit network, the ICMP will report that problem.

**Internet Protocol Version 4 (IPv4)**

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IPv4 is 32-bit addressing scheme used as TCP/IP host addressing mechanism. IP addressing enables every host on the TCP/IP network to be uniquely identifiable. IPv4 provides hierarchical addressing scheme which enables it to divide the network into sub-networks, each with well-defined number of hosts. IP addresses are divided into many categories:

- **Class A:** It uses first octet for network addresses and last three octets for host addressing.
- **Class B:** It uses first two octets for network addresses and last two for host addressing.
- **Class C:** It uses first three octets for network addresses and last one for host addressing.
- **Class D:** It provides flat IP addressing scheme in contrast to hierarchical structure for above three.
- **Class E:** It is used as experimental.

IPv4 also has well-defined address spaces to be used as private addresses (not routable on internet), and public addresses (provided by ISPs and are routable on internet). Though IP is not reliable one; it provides 'Best-Effort-Delivery' mechanism.

**Internet Protocol Version 6 (IPv6)**

Exhaustion of IPv4 addresses gave birth to a next generation Internet Protocol version 6. IPv6 addresses its nodes with 128-bit wide address providing plenty of address space for future to be used on entire planet or beyond. IPv6 has introduced Anycast addressing but has removed the concept of broadcasting. IPv6 enables devices to self-acquire an IPv6 address and communicate within that subnet.

This auto-configuration removes the dependability of Dynamic Host Configuration Protocol (DHCP) servers. This way, even if the DHCP server on that subnet is down, the hosts

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can communicate with each other. IPv6 provides new feature of IPv6 mobility. Mobile IPv6-equipped machines can roam around without the need of changing their IP addresses. IPv6 is still in transition phase and is expected to replace IPv4 completely in coming years. At present, there are few networks which are running on IPv6. There are some transition mechanisms available for IPv6-enabled networks to speak and roam around different networks easily on IPv4. These are:

- Dual stack implementation
- Tunneling
- NAT-PT

**Real-time Transport Protocol**

One of the Internet protocols that can be used in conjunction with reservation models at the network layer is the Real-time Transport Protocol (RTP) [RFC1889]. RTP is an end-to-end protocol for the transport of real-time data. An important application type supported by RTP is multi-party conferencing because of its support for synchronization, framing, encryption, timing and source identification. RTP has its companion RTP Control Protocol (RTCP), which is used to interchange QoS and failure information between the QoS monitor applications in the end-systems.

RTP does not define any kind of QoS itself and does not provide re-ordering or retransmission of lost packets. However, it provides a sequence number that enables the application using RTP to initiate such steps. RTP is typically used directly on top of UDP/IP or on top of ST-2. In the former case, QoS can be guaranteed by the use of RSVP's reservation mechanisms for the UDP datagrams. In such a combination, the RTP stack provides the information necessary to make educated guesses about the behavior of the data stream based on RTP's knowledge of the data format. In addition to the base RTP specification, a number of companion documents exists that provide encapsulations for various continuous media formats such as M-JPEG or MPEG. Hence, RTP itself provides no real QoS support; it relies on other appropriate protocols and mechanisms.

The IPv6 network showed a better quality, as compared to the IPv4 network. Also the bandwidth consumption of the IPv6 network is much more stable as compared to the IPv4 network. Also, the UDP foreground traffic, Ex. multimedia traffic, suffers less packet loss with respect to the TCP background traffic (generated artificially through MGEN Traffic Generator Tool).

The results derived after the experiment was used to display packet loss, delay and latency as shown in Figure 3. The MPEG-2 TS (Transport Stream) dissector for Ethereal can display transport stream and section heading information, reassemble the sections and analyze the PSI (Program Specific Information) data. Open the .ts file (created through VideoLAN), from Ethereal (version 0.10.8).

There are two possibilities to dissect MPEG 2 ULE (Unidirectional Lightweight Encapsulation), we will use etherial "Enable MPEG2 ULE SNDU analyzation" option as shown in Figure 4. If these settings are wrong the analysis will be incorrect. The Figure 4 shows the Transport priority marked with red circle. The experimental results indicated that priority based IPv6 traffic has less of packet loss, latency and jitter problems as compared to the normal IPV6/IPv4 network.

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**Routing**

QoS driven routing algorithms are needed for the efficient establishment of reservations. These algorithms suggest one or multiple suitable paths towards a given target considering a given set of QoS requirements. Then one attempts to make a reservation on such a path. Without appropriate routing mechanisms which take QoS requirements into account, the setup of reservations becomes a mere trial-and-error approach.

A QoS driven routing algorithm has to consider the currently available capacity of a resource to avoid an immediate rejection of the reservation attempt and the QoS requirements of the reservation to find a route best-suited for this QoS. It should also consider the resource load after the routing decision to avoid using up the majority of resources on this route. Some of the problems to be solved with QoS routing are: how much state information should be exchanged among the routers; how often should this state information be updated; must there be a distinction between exterior and interior systems and if yes, how can it be made; is it possible to hide internal details of an autonomous system; can the complexity of path computation be managed? QoS routing is currently still in its infancy.

At least in the Internet, its necessity, and the ability in principle to perform QoS routing and the proposed approaches are currently under controversial discussions. Furthermore, it seems difficult to combine QoS routing and receiver-oriented reservations. The hard state, sender-oriented ATM camp, on the other hand, designed PNNI which provides at least some QoS routing support.

**III- Conclusion**

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The complete architecture and configurations was considered more preferable due to its extensibility features, powerful API, event driven approach etc. The proposed module implemented to gives the priority to multimedia traffic gives significant improvement in multimedia streaming application. Table 2 conforms that if we can release router to priority of Multimedia Streaming data through kernel level IPV6 field modification then the user can get waste differentiated performance and QoS parameter optimization in the output. Also, the experimental results indicated that priority based IPv6 traffic has less of packet loss, latency and jitter problems as compared to the normal IPV6/IPv4 network.

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