"Odysseus-2001"

Media PEP

(Media Performance Enhanced Proxy)

A Scaleable and Dependable
ActiveUMTS QoS Management Server,
An Internet Protocol Booster and
An Adaptive Media Transcoder Switch Architecture for
E2E-IP Integration of Mobile Wireless Interactive
Telepresence Applications

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1. Introduction

There are two clear tendencies in the telecommunication market:

- the *customers' raised expectations* on the efficiency of the access to a great number of information sources and services.
- the *customers' demand on the mobile access* to information and services.

It is expected to address these targets by the further development and the integration of Internet and mobile wireless networks. With the UMTS, the 3rd generation of global mobile networks, a convergence of multimedia applications and the mobile communication world is going to be achieved.

By the initiation of mobile multimedia services, including the mobile Internet access, high technical demands arise on the underlying systems. For UMTS the support of the Internet and of multimedia services with QoS guarantees is very important. Two essential technologies are foreseen in UMTS: W-CDMA and TD-CDMA for the radio access interfaces and IP connectivity in UTRAN.

The end-to-end QoS is the main topic in the discussion about the service availability. However, QoS does not solely depend on the network performance, but also on the capability of networks and the inter-systems together. Therefore, we have to consider these two QoS factors from the viewpoint of the UMTS in our concept.

The target of our project is the development of a dynamic concept for QoS management in UMTS communications. This concept integrates adaptive QoS control mechanisms and algorithms for the automatic bandwidth management in combination with the customers' profiles and their feedback during a running multimedia conferencing session.

2. Quality of Service Management in UMTS

In traditional networks, the customer specifies distinct *static QoS* requirements. Then, a traffic contract is agreed upon between the customer and the network after connection establishment and previously to data exchange. The data stream must be conformed to the parameters specified by the customer in order to obtain the required QoS guarantee.

In mobile communications we intend to modify this process. It is more useful if the customer can *dynamically influence the QoS parameters* in accordance with her actual preferences during data exchange. In UMTS the application provides the possibility to control the parameters themselves through a feedback channel. Two consequences result from this:

- QoS-aware applications and
- programmable networks

The QoS-requirements should be considered from two viewpoints:

2.1. Network's point of view:

- QoS mechanisms should allow the efficient use of radio resources.
- QoS parameters should support different levels of QoS requirements.
- QoS mechanisms should allow the independent evolution of core networks, access-networks and UMTS networks.

2.2. End-User's point of view:

- QoS should be made available on an end-to-end base.
- The end-user can define traffic parameters and control them.
- The customer interface for this control should be simple, with only a few parameters that can provide the required feedback information to the network upon simple interactions.
- applications in UMTS have asymmetric and symmetric behaviour between uplink and downlink. The QoS attributes should support both directions.

3. Survey of the Quality of Service Structures

The previous efforts for structures of QoS can be classified in the following manner:

- End systems (operating system, scheduling in the end systems, distributed-platform)
- Transport systems (transport service and transport protocols)
- Internetworking (flow control, congestion control, service discipline, traffic management)

There are different concepts and mechanisms for QoS management and preservation used at the moment. In particular, IntServ, DiffServ, MPLS, and the Bandwidth Broker are quite popular. *All these concepts were designed for fixed networks.*

Among the previous partners' work on the issues of QoS guarantees two particular approaches are worth to be mentioned. The first one is the <u>Interval-based Real-Time Data Transport</u> and the second one is the <u>Adaptive Bandwidth Management.</u> The later is characterized by a combination of various priority types in bandwidth scheduling and an integration of scheduling, flow control and admission control for ATM networks. For the guarantee of QoS the only requirement is that the source traffic satisfies the leaky bucket constraint.

4. A Framework for QoS Guarantees in UMTS Networks

We believe that a complete solution to guarantee end-to-end QoS can result only by involving the key network components with respect to all layers of the protocol stack and consider the interactions between them. In this relation, it is to be considered that not only the inter-systems, but also the end systems impact the traffic behaviour. In the following, we will describe our framework for QoS guarantees in some detail.

4.1. Concept of Application-Aware QoS Management

The following factors will be considered in the design of a new concept for application-aware QoS management :

- Wireless communication has special qualities compared to traditional wired communication, particularly because of:

    - Deficient resources,
    - Limited bandwidths in wireless links,
    - Failing transfer channels,
    - Limited energy supply.

- Multimedia applications (in particular novel applications) for wireless communication require a *higher performance* of the networks.

- Traditional protocol structures (IP) are based on the layer concept. There is no support for guaranteeing QoS between the layers.

- The new protocol versions should be considered with respect to the features of the wireless networks. In particular, the limited performance of the current mobile terminal equipment has to be taken into consideration.

- Supporting the development of new multimedia applications is of crucial importance for UMTS.

Therefore, in the first phase of this project we are going to examine and specify exactly the requirements for QoS guarantees in UMTS and define an abstract and aggregated dynamically negotiable QoS concept.

In the application-aware context, it is necessary to consider the user's requirements regarding the availability of service quality provided by the network. Therefore, the QoS management is in charge of accessing essential information about the offered service whether it satisfies the user's requirements.

By taking into account the ETSI 3GPP recommendations for UMTS, an abstract and aggregated dynamically negotiable concept for QoS guarantees will be defined from the viewpoint of an application-aware UMTS architecture. The concept will be analysed and compared to other existing concepts and mechanisms of the standard Internet and ATM traffic models. We expect to provide some traffic monitoring benchmarks for guaranteeing QoS in UMTS environments.

In the design of the QoS concept, it is essential to consider the mechanisms for the resource reservation, resource negotiation and resource management at different service levels. The impact of the current RSVP protocol and MAC protocol is going to be analysed and evaluated for wireless networks. The MAC layer has direct impact on the availability of QoS guarantees. In this relation, it required that an information exchange between the MAC layer and the application layer takes place. Using the information from the MAC layer, the application can adapt to the fluctuations of the access network and adjust its service quality. On the other hand, the application can also enforce its QoS requirements dynamically to the MAC layer.

This is a way for enabling the re-negotiation during the session which is to be expected in UMTS multimedia communications. Another approach to do this is to extend the available protocols (such as RSVP). Research in this area is out of the scope of this work.

4.2. Design and Implementation of an Active UMTS QoS Management Server

Another part of this project considers the design and implementation of an *Active UMTS* QoS Management Server. With this concept, a simple support for the customer control and the possibility for a rapid deployment of new services can be implemented. A QoS-based API structure should be provided.

The heterogeneity of UMTS networks will be also taken into consideration, e.g. various bandwidths and error rates along with different terminal performances during the media exchange between handhelds, laptops and desktop computers.

There are different user preferences. The QoS requirements of the various customers can be very different. Within UMTS the applications should be able to control their QoS. The QoS Management Server should realize the QoS co-ordination between the customer and the network and devise the best possible strategy to for providing an aggregated QoS. This what we call *active network and user feedback*.

The Active UMTS QoS Management Server has the following functions:

- User-profile definition and specification (corresponding to different user classes: Gold Card User, Premium Card User, Surfing User).
- Policing function: Policing corresponding to the different user profiles.
- QoS admission mechanisms.
- Mapping function: Translation of the user QoS requirements in net parameters and vice versa.
- Control function: dynamic distribution of the RAN resources (allocation).
- Monitoring: controlling whether the QoS contract can be hold on allocated resources (RAN). On the other hand, a dynamic re-negotiation of QoS parameters is going to be carried out by means of interlayer feedback channels.

The QoS Management Server is based on the client server model.

5. Boosting Wireless Performance

Real-time wireless mobile multimedia telecommunication is going public in 2002 with worldwide wireless networks like UMTS and IMT-2000. The basic premises are the increased available bandwidth (up to 2 Mbit/s) compared to today's circuit switched GSM (9.6 kbit/s) and the efficient media coding techniques such as MPEG-4. Unfortunately, the available bandwidth is going to oscillate during a multimedia conferencing session because of the characteristics of the wireless link such as packet losses and delay jitter resulting from field interferences, handover and disturbing Internet traffic.

These effects inevitably lead to unacceptable Quality of Service degradations. Although MPEG-4 coding is a DCT compression technique, which can adapt to the available bandwidth and thus efficiently use the frequency bands (e.g. by spatial scalability using multiple compression streams), the encoder at the transmitter side requires knowledge about the available bandwidth. In circuit switched wireless networks this bandwidth (and QoS) is fixed and limited to voice and simple data query services based on WAP. In a packet switched wireless network such as UMTS, we have larger bandwidth enabling multimedia, but also high traffic and unstable wireless channel characteristics. Therefore, a *feedback channel* and a *stabilization of bandwidth* per stream and per user is required to guarantee a desired level of QoS. But, how this can be provided in an effective manner to avoid latency and processing overhead ?

One possible solution can be achieved by special *data link protocols adapted to specific MPEG encoders*. For instance, in MPEG-2 there are some packets belonging to the so called B-frames which contain redundant information, and other packets belonging to the important I-frames containing the essential code change in a video stream. What is required from the network is, that in case of congestion, the data link protocol selectively discards the assembled packets based on the information about their type. In addition, the data link protocol may also integrate some reservation algorithms to guarantee QoS requirements.

The wireless access is characterized by a completely different behavior of the network when compared to the wired Internet. Thus, the main reason for QoS degradations is that the existing Internet traffic control mechanisms such as TCP and UDP were primarily developed for the use in wired networks. Therefore, the wireless part of the packet network has to be treated separately. For instance, the Wireless Access Protocol (WAP) is such an approach. However, it is completely orthogonal to the straightforward end-to-end IP QoS concept. WAP breaks the IP net in two parts and requires a complete exchange of the Internet protocol suite and applications.

In contrast, other approaches does not require re-implemetation of existing protocols and applications. Protocol booster architectures, also called Performance Enhancing Proxies (PEP), integrate performance-enhancing functionality that can be located at the edges of the wireless part of the network. These protocol boosters operate *transparently* without the need to modify the existing IP suite. To enable efficient operation the boosters have to be designed for specific applications. For the case of TCP applications IP booster architectures can double TCP throughput even under noise propagation conditions. Real-time video conferencing applications require other specific booster functionalities.

The *"Odysseus-2001"* project is an initiative of Siemens AG, TU Berlin, the Heinrich Hertz Institute (Berlin) and TU Ilmenau. One of its targets is the development and the demonstration of a data link protocol which integrates IP booster mechanisms following a UMTS conform dynamic QoS model and algorithms for adaptive bandwidth management of user profiles in UMTS multimedia communications and fits to the needs of the MPEG-4 coder to improve the video conferencing quality within the *HiCell* m-Commerce application scenario. The data link protocol can be integrated in the end-system and base station (Radio Node System, RNS), but may be also part of a proxy architecture enabling code propagation and installation on demand.

6. The UMTS Video Conferencing Demonstrator

Although UMTS technology is not available for experiments at the moment, the potential lying in UMTS can be demonstrated by using similar wireless technologies. One of the candidates is the IEEE 802.11 wireless LAN (WLAN) network, which offers transmission speeds up to 11 Mbit/s.

The *"Odysseus-2001"* project is going to develop a demonstrator for video conferencing using IEEE 802.11 wireless LAN technology. The demonstrator is the basis for *measurements* to investigate the potentialities and the limits of video conferencing in today's wireless networks. The Quality of Service for the *HiCell* video conferencing application is going to be measured and evaluated using standard QoS parameters like packet losses, delay and jitter.

In parallel, an active network methodology for dynamic and adaptive bandwidth management is going to be developed. This methodology will be based on the analysis of the standardized QoS concepts, architectures and performance characteristics of UMTS (ETSI, 3GPP). This work will show the differences of today's video conferencing requirements and functionality that is needed in future. The results of this work will be further analyzed by integrating and testing the "backward QoS channel " data link protocol and the MPEG-4 codec via an appropriate bit rate management API into the end-to-end trial system. Finally, a method for applying application-aware QoS management for UMTS using *dynamic in-band signaling* and *adaptive bandwidth allocation* along with standard resource reservation mechanisms within OSI layers 2 and 3 will be proposed and implemented.

Parts of the demonstrator are Linux Laptops equipped with a wireless LAN interface from Lucent Technologies. For video conferencing MINT – a multimedia tool – developed by GMD Fokus and TU Berlin will be used. MINT integrates public domain multimedia tools (vic, vat and whiteboard) via a user friendly graphical user interface. Participants are addressed by their e-mail address. A location server redirects connection attempts to the host, where the addressed user is currently logged in.

The *HiCell* m-Commerce demonstrator scenario is organized as follows:

One or multiple IEEE 802.11 wireless radio cells are connected via Ethernet, which in turn is connected to an Internet backbone. Multiple end-systems located in the radio cells are equipped with video cameras, headsets (or separate loud speakers and microphones). Using the multimedia tool MINT, a session chairman can invite other users to participate to a video conference via their e-mail addresses. A location server is searching the end-systems, where the participants are currently logged in and forwards the conference request to them. Then, the mobile end-system is ringing and the requested user may accept or deny the call. Upon acceptance, the MINT client opens a conference window, where all the participants are visible in small windows with a real life video. By clicking into the small windows, the window size can be raised. During conversation, users can freely move within the radio cell and even change the radio cell. If the user moves into a "radio hole", the application will reduce the bandwidth automatically to adapt to the changing characteristics. The entire equipment is installed on laptops, wearable PCs or desktop computers running the Linux operating system.

7. Integrating MPEG-4 into the MINT video conferencing system

Investigating the behavior of MPEG-4 in wireless Internet is one major target of the “Odysseus-2001” project. For this reason the demonstrator is going to be extended by an MPEG-4 transcoder switch which is going to support wireless video conferencing scenarios with assumes the MINT multimedia tool. In addition, the real-time MPEG-4 bit rate adaptation mechanisms are going to be integrated within the Real-time Transport Protocol (RTP) and the Session Invitation Protocol (SIP) to allow bandwidth reservation via RSVP.

Furthermore, the performance of the MPEG-4 sessions is going to be measured and compared with the previous results of wireless video conferencing without dynamic bandwidth adaptation. It is expected, that the Quality if Service will be improved. Finally, if necessary, the system will be tuned by optimizing parameters such as packet size and required bandwidth and including additional feedback information into the Real-time Transport Control Protocol (RTCP) as required by the MPEG-4 transcoder switch.

8. Development of MPEG-4 adaptive data link protocol mechanisms

Up to this point we are going to obtain knowledge about the QoS of current underlying transmission systems and of future MPEG-4 based video conferencing systems without modifications of the protocol stack of the end-system or base station to match the MPEG-4 encoding.

It is expected, that the Quality of Service can be improved by additional protocol mechanisms integrated into a data link protocol of the wireless part of the network for the following reasons:

- First, the feedback information between the end-systems via RTCP reflects *only the end-to-end status* of the network, but not the rapid change of the wireless link characteristics. This means that the end-to-end network information may irrelevant and the bit rate adaptation of the MPEG-4 transcoder may be wrong.

- Second, the adaptation area of the MPEG-4 transcoder is limited. Even if the coder obtains knowledge about wireless link characteristics, it could not adapt fast enough to the changes. In other words, the wireless part of the network will dominate the end-to-end QoS characteristics.

- Third, due to the high data compression in MPEG-4 managing congestion control in the traditional way does not seem to be the appropriate; the loss of a single packet belonging to an I-frame can significantly reduce the QoS of the video stream. Therefore, a code-oriented protocol mechanism is required in order to selectively drop only the “unimportant” packets.

In the “Odysseus-2001” project we are going to specify, develop and implement a data link protocol which alleviates the above problems. It is going to reside on top of the wireless technology dependent part (either IEEE 802.11 or a CDMA based system like UMTS) and will be technology independent. In a second step the protocol will be adapted to CDMA systems like UMTS.

The data link protocol is going to stabilize the important QoS parameters (packet losses and delay jitter) by introducing a play-out buffer and additional ARQ based error correction mechanisms. In addition, multiplexing (e.g. a single conference stream or multiple conference streams per connection, separation of audio and video streams), segmentation and reassembling mechanisms (e.g. combining multiple packets into a single MAC packet) are going to be provided. The play-out buffer will delay packets at the receiver for some short time to guarantee a continuous data stream without jitter at the receiving mobile end-system. In this way the probability that no data is available for the decoder due to great delay variations of the wireless link will be minimized. The play-out buffer has a small play-out delay to enable interactive communication and spare memory resources. To ensure that the play-out buffer contains nearly always data, the number of retransmissions introduced by the ARQ protocol must be limited. (Too many retransmissions introduce jitter which cannot be compensated by the play-out buffer.) The number of retransmissions will be adapted to the number of packets residing in the play-out buffer, which is in turn dependent from the loss rate of the wireless link and the elasticity of the MPEG-4 transcoder. This algorithm can be integrated in each data link protocol independently from the preferred wireless transmission technology. However, the choice of the preferred play-out delay and the optimized packet sizes will be technology and application dependent.

In the second year of this project we will focus on the design and the implementation of a MPEG-4 protocol booster located in the data link layer. Protocol boosters/proxies transparently (w.r.t. IP) improve the QoS like chemical catalysators. This means that higher layer protocols and applications do not recognize the additional functionality provided by that network agent. The booster is going to look into the MPEG-4 stream and mark packets, which are content sensitive and which *must be transferred error free* to the receiver. Otherwise the QoS may degrade in an unacceptable way. To ensure that the marked packets are going to be transferred correctly, the following mechanisms are going to be implemented into the data link protocol and evaluated with respect to their performance:

1. Irrespective whether the play-out buffer becomes empty and additional jitter is going to be introduced, the sender will continuously retransmit the packet and deliver it to the receiver. To compensate this additional delay, the sender will discard packets which follow with lower priority. As a result only a small buffer at the sender side is needed. Regardless of the loss of a few unimportant packets, the receiver can recover the lost information.

2. If more buffer space is available at the sender, subsequent packets (regardless of their relevance) will be stored during the re-transmission of important packets. Only in case that the sender buffer overflows, unimportant packets will be discarded. It is expected that this algorithm will reduce information loss, but will introduces more jitter compared to the first algorithm.

Independently which algorithm is going to be used, the maximum number of retransmissions must be limited. To find an appropriated value is also subject of this project.

In addition, the data link protocol is going to provide the MPEG-4 transcoder switch with the currently valid information about the wireless link. This will include general information such as the available bandwidth, loss rate or jitter, but also some specific parameters required by the MPEG-4. The specification of the relevant parameters and the mapping of the network parameters into parameters understandable by the MPEG-4 coder is subject of this project. Also, the implementation of the signaling mechanisms between the data link layer and the MPEG-4 coder are going to be investigated in this work. Results could be delivered to the Siemens standardization efforts to integrate the MPEG-4 specific control functions into RTP.

The adaptive transcoding mechanisms for wireless LANs will be integrated into the demonstrator. The achievable performance is going to be measured and evaluated. The data link protocol for UMTS will be modeled and evaluated via simulations. The algorithms will be based on the SMPT algorithm of the TU Berlin [IEEE Communication Magazine (Vol. 38 Oct 2000), Special issue on G3 wireless systems.] The basic algorithm of SMPT has the objective to reduce the loss rate by a fixed preset jitter bound. SMPT assumes that it is possible for a single end-system to acquire multiple CDMA codes. Normally, the end-system is using only use one code. In case of error packets, corrupted packets will be retransmitted using another code, while subsequent packets are transmitted in parallel with the old code. At the end of a retransmission code the resulting jitter for the error free packets and the retransmitted packets will not be recognized by upper layer protocols or applications. Only if no more codes for retransmissions are available, the packet is going to be discarded. To obtain information about successful delivery of a packet a back channel is required. This algorithm will be expanded for the specific use in MPEG-4. Basically, the algorithms described above will be adapted to SMPT.

9. Integration and test of the *ActiveUMTS QoS Management* methods into MINT and the adaptive DLL protocol mechanisms

The HiCell video conferencing demonstrator is going to be integrated and tested in the Siemens FutureLab in Berlin with the following objectives:

- Integration of new and updated Internet protocol versions (like SIP),
- Improving the stability of the demonstrator
- development of a user-friendly graphical user interface suitable for demonstrations.
- System test of the new developed protocol features (MPEG-4 coder and data link protocol) integrated in the Active QoS Management System.

10. Adaptable Bandwidth Error Resilient MPEG-4 Transcoding Techniques

MPEG-4-compressed video signals are extremely vulnerable to transmission errors. Appropriate channel coding techniques can reduce transmission errors. For channels without memory, such as the AWGN channel, channel coding techniques provide very significant reductions of transmission errors at a comparably moderate bit-rate overhead. For the mobile fading channel, however, the effective use of forward error correction is limited when assuming a small end-to-end delay. Here the use of error resilience techniques in the source codec becomes important.

When operating a hybrid video coder in prediction mode, the loss of information in one frame has a considerable impact on the quality of the following frames. As a result, spatio-temporal error propagation is a typical transmission error effect for predictive coding. Because errors remain visible for a longer period of time, the resulting artifacts are particularly annoying to end users. To some extent, the impairment caused by transmission errors decays over time due to leakage in the prediction loop. However, the leakage in standardized video decoders like MPEG-4 is not very strong, and good video quality can only be achieved when

1. spatio-temporal error propagation is completely avoided or
2. the impact of spatio-temporal error propagation is mitigated.

Both approaches are being investigated in this project.

The first approach, avoidance of spatio-temporal error propagation, is addressed via temporal and spatial scalable coding. For both techniques, higher resolution representations of the video signal are added as enhancement layers to a base layer representation. The base layer can be correctly decoded without knowledge of the enhancement layers. This decoupling of the enhancement layers from the base layer permits the use of unequal error protection (provided by TUB) so that the base layer will most at the time be correctly decoded. If additionally, the transmission conditions are suitable for higher bit-rates, the end-user can be provided with the higher resolution enhancement layers.

The second approach, mitigation of the impact of spatio-temporal error propagation, is addressed via the following techniques:

- Error concealment
- Syntax scalability
- Intra coding with and without feedback
- Reference pictures selection with feedback.

Syntax scalability is achieved via the MPEG-4 data partitioning mode which separates important and less important information of a picture. In case only the important information is received (using unequal error protection), an erroneous version of the picture can still be recovered. This information can be utilized to support error concealment to obtain a good approximation of the reference picture at the encoder side. This scenario will cause spatio-temporal error propagation but provides a further level of scalability that can be used to display a still acceptable version of the video signal.

The impact of the spatio-temporal error propagation can be reduced using Intra coding and reference picture selection. Intra coding can be employed with and without utilizing feedback, while the use of reference picture selection requires feedback in MPEG-4.

The *workplan* of the project includes first an investigation of the efficiency of the various approaches and than a real-time implementation of the considered techniques. The efficiency evaluation (considering the rate-distortion to complexity trade-off) is conducted via specifying interfaces and statistics of the transmission channel.

Two scenarios of scalable coding are being investigated:
1. Scalable coding at the far-end encoder without feedback messages
2. Scalable transcoding including feedback messages

The efficiency evaluation for scalable coding at the far-end encoder without feedback messages is planned for the following techniques
- Temporal scalability
- Spatial scalability
- Syntax scalability
- Error concealment
- Intra coding without feedback

The efficiency evaluation for scalable transcoding including feedback messages includes
- Bit-rate conversion
- Format conversion (MPEG-2/4 and H.263 to MPEG-4)
- Intra coding with feedback
- Reference picture selection with feedback.

Based on the evaluation of the various approaches, a design decision is made for the most efficient techniques and those are realized as real-time implementations and are integrated into a transmission system to obtain a real-time demonstrator.

11. High-Performance Real-Time Distributed Computing for 3G Mobile Internet

The popularity of Internet is now making possible to deliver computing services to millions of households and individuals worldwide, provided that the networks and the software can meet the challenge. One of the major challenges in the area of Network Computing is creating reliable distributed systems that are based on commercial computing and communications technologies.

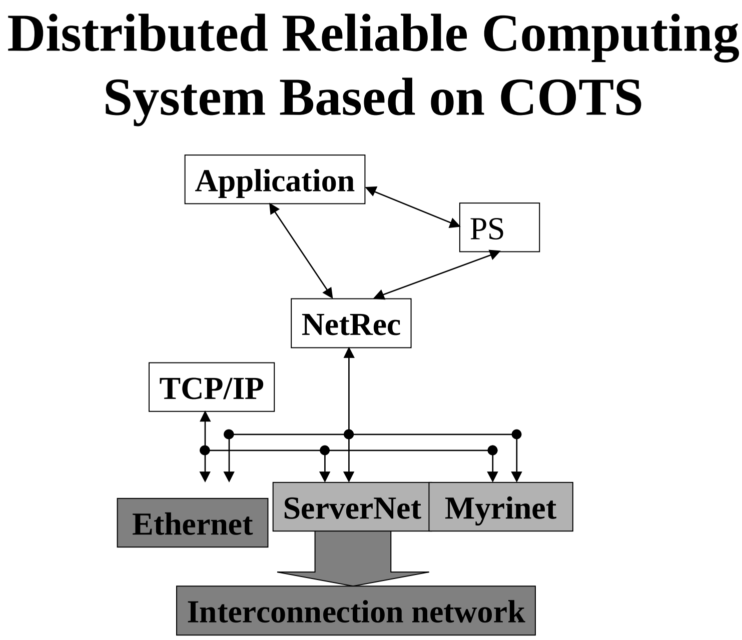

Fig.1 : A Distributed Reliable Computing System based on COTS

The architecture in Fig.1 represents a reliable heterogeneous distributed computing system based on inexpensive Components off-the-Shelf (COTS). Each of the nodes is a standard personal computer or workstation, using a standard operating system, such as Linux or Windows NT/2000. The nodes can be connected via legacy switches and adapter cards to build up different point-to-point networks. In particular, the *ServerNet* architecture offers a simple, comprehensive solution for building such System Area Networks (SAN) that meet the demanding requirements of high availability and scalability. In addition, the single nodes can be connected by a fast Ethernet or Gigabit Ethernet using the standard protocol TCP/IP.

Thus, by using a special software for creating *virtual topologies*, any user application such as Stream Multimedia and Real-time Video Conference Service, can be mapped on the distributed reliable computation system.

The execution time as a function of the compression ratio for a quad-tree data compression program executed in a given topology is presented in Fig.2 for the following operational modes: single node, Ethernet using TCP/IP and ServerNet1.

Quad-tree Compression Algorithm (compression ratio vs. time)

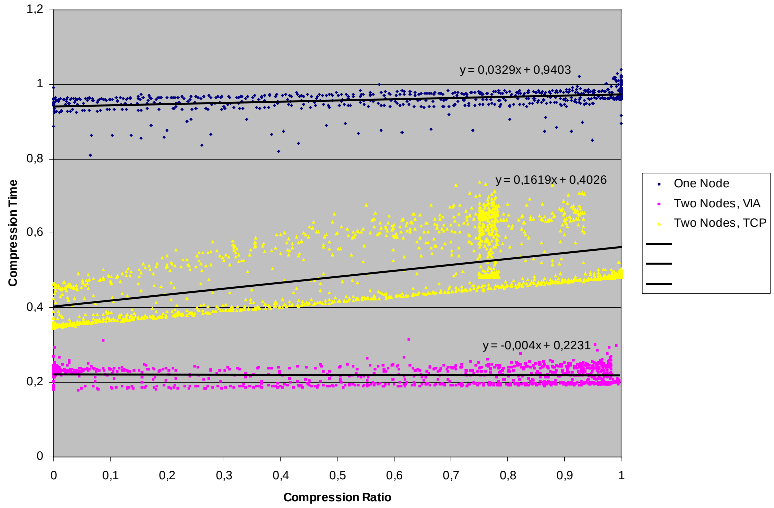

Fig. 2: The Execution of a Quad-tree Compression Algorithm

As expected, the execution times over ServerNet -II are smallest because of the advanced features of the ServerNet -II . The ServerNet cluster topology will allow multiple streams to be distributed among processing nodes and to be processed in parallel. By increasing the number of the users, the ServerNet cluster can be scaled and by increasing the size of the topology the QoS requirements can be satisfied. The communication pattern of the algorithms for Stream Multimedia and Real-time Video Conference Service will be used to select the best topology to provide high throughput and low latencies. In this way, the QoS requirements for these types of algorithms will be met.

12. A Scaleable and Dependable Server for Stream Multimedia and Real-Time Video Conferencing Service

A scalable and dependable server for Stream Multimedia and Real-time Video Conference Service will have the following characteristics:

- A **deadlock and livelock free** architecture
- **Scalable architecture** providing *low latency transfer*: as the network size increases, the throughput increases linearly, while the increase of the average latency is smaller.
- Based on **standard** hardware **COTS** (Components Of The Shelf) and software modules (routers, drivers, etc.): Linux, (Windows NT/2000 based)+ Message Passing Interface .
- **adaptable routing** algorithms acc. to the very heavy congested or faulty links and nodes
- **predictable delays** acc. to the max. delivery times inside the cluster
- **application oriented QoS** management – dynamic bandwidth allocation and scalable/ adaptable streaming
- Support of **multicast and broadcast** modes inside the topology

This features will allow the size of the video server to scale depending on the number of the users ranging from 10.000 (at the first step) up to 1.000.000 at the end of the project. The server will provide an efficient implementation of the scalable video and QoS Management techniques. The topology of the server will be selected based on the communication pattern of the software packages developed and used by the partners in the project. This will require the collaboration with the partners to start at the beginning of the project.