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# IP Videoconferencing using a Quality of Service Public Network

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**Abstract**

*Videoconferencing in the medical world has been successfully used for quite a few years. Nevertheless it has not spread significantly in daily use. Some of the problems rely on the infrastructure needed to set up a video session: one or more ISDN lines or a fast Internet connection. The first is not easily available everywhere in a building; the latter is rarely so fast to allow for a smooth operation with no quality drops. The use of Videonet, the first European commercial public Internet link with guaranteed Quality of Service (QoS), has the potential to be a breakthrough in videoconferencing. We describe this new system and its applications, with the first tests in a hospital environment. Our results show that there are still problems to be solved in order to achieve a quality comparable to ISDN.*

***Keywords:***

Teleconference; Telemedicine; Videoconferencing; Quality of Service; Telecommunications

## Introduction

Videoconferencing in the medical world has been successfully used for quite a few years. Applications range from simple communication to training [1] and teleconsultation. Even though there are a number of studies regarding the positive effect of videoconferencing (reducing travels, saving money, connecting disabled persons, accessing far away specialists) it has not spread significantly in hospital daily use. The first really usable systems were available on the market when ISDN lines were installed. The digital quality of this kind of digital telephone connection gave more stability to the video sessions. An *umbrella* standard was developed, called H.320 [2] and all hardware and software makers complied with it in order to guarantee interoperability. It coordinates other protocols for different purposes, providing the mechanism of negotiation among users of audio, video and application sharing tools.

In order to start a videoconference using H.320, one or more ISDN lines are required. Typically this is easy to accomplish in a specific place, but very rarely there are buildings, like hospitals, with ISDN plugs in all rooms. The cost of a videoconference with an ISDN-2 connection at 128 Kbps is double the cost of a plain telephone call. Quality is sufficient for a visual dialogue. For a high quality teleconsultation three ISDN-2 lines are normally required, which means six times the cost of a phone call. If the call is in town, the cost is low: when international partners are involved, is may be very high, although lower than flying to the remote place. The audio and video quality connection is steady because all the bandwidth is completely reserved to the purpose. Dropped lines are noticed from time to time, mainly on international multiconferencing calls, due to some differences among the national implementations of the standard: we verified this malfunction in the TaRgET [3] project on the EURO-ISDN network.

The diffusion of the Internet has led many people to start using it also for audio and video calls. The IP infrastructure was not meant for real-time applications and suffers from delays and bandwidth restriction problems. The H.323 [4] protocol addresses videoconferencing over IP connections. Quality is not guaranteed and therefore the audio and video flow can degrade or stop when there is other heavy traffic over the same lines. A quote can show how sometimes the problem is handled: "*Our network is quite fast, we can use the 10 Mbps bit rate and in this network there are no other activities in this moment, so there are not any interferences*" [5]. Obviously this is not a normal scenario.

An alternate solution to optimize resources and to efficiently distribute the videoconferencing facilities on an IP network is to make use of a H.323 to H.320 gateway. Starting the call from any IP connected computer on the local network, the connection to the distant partner can be established using the ISDN line through the gateway. The stable quality of H.320 merges with the versatility of the H.323. The price is high because of the increased complexity and the ISDN transmission costs.

The ideal scenario is a guaranteed Quality of Service (QoS) transmission over the public Internet in order to ensure a steady flow of audio and video. After many researches, commercial offerings have just begun. In the "Campus Bio-Medico" University Hospital of Rome, and in the Department of New Technology of ELIS Consortium, in cooperation with Telecom Italia, Cisco Systems and HP, we have pioneered Videonet, the first QoS public network in Europe.

## Materials and Methods

### IP Videoconferencing

The universal use of the Internet Protocol (IP) in all kind of applications makes it a preferred solution also for audio and video connections despite quality problems and difficulties in real-time response. Specific protocols have been developed to solve part of these problems. The largest interest of the business community is on Voice over IP (VoIP) because it can enormously reduce telephone bills. The cost/benefit ratio of an IP solution is very favorable due to the large amount of users who can be involved.

IP videoconferencing is different from IP streaming of audiovisual content because it requires real time interaction among two or a few participants. Streaming is for broadcasting to many, even millions of users, with almost no feedback and with no interaction. When streaming content, it is easier to address quality problems because a partial buffering can be performed to deal with possible packet losses. This means that there are some seconds of delay between transmitting and receiving and that the Round Trip Time (RTT) of the IP packets is not crucial.

### Quality of Service

Quality of Service (QoS) is generally intended as the quantitative and qualitative characteristics needed to achieve the required functionality of an application, including the user's satisfaction. QoS parameters may be oriented towards:

- *performance* - sequential versus parallel processing, delays, data rate;
- *format* - transfer rate, data format, compression schema, image resolution;
- *synchronization* - loosely versus tightly coupled, synchronous versus asynchronous;
- *cost* - platform rates, copyright fees, connection and data transmission rates;
- *user* - subjective quality of images, sound, response time.

The OSI Reference Model has a number of QoS parameters describing the speed and reliability of transmission, such as throughput, transit delay, packet latency, error rate and connection establishment failure probability. Standard Internet Protocol networks provide “best effort” data delivery by default: the IP protocol concerns only about the routing of the packets, that is the way they can reach their destination. Using another protocol over IP (typically TCP), we are sure that data is delivered, but we don’t know how long it takes to reach destination. This is a simple framework, which scales well and allows rapid growth as it happens in the Internet. Service is not denied if traffic grows: things only tend to be slower. The delivery delays are variable (the so called *jitter*) and packet loss is not predictable: they do not greatly affect email, file transfer and web browsing. On the contrary, multimedia and real-time applications have very strict requirements:

- *bandwidth*, critical for real time video transmission;
- *synchronization* among different media flows (audio, video, data);
- *low jitter* for keeping a steady flow with fixed delays;
- *timeliness* for effective interaction in two-way applications like telephony or videoconferencing.

In order to provide QoS in the Internet, some mechanisms need to be deployed. We describe some of them.

#### *Increasing Bandwidth*

The easiest way to provide QoS is to increase bandwidth and resources so that all the traffic receives a good service. Some experts think that fibre cabling and *dense wavelength division multiplexing* (DWDM) technologies will make bandwidth so abundant and cheap that high quality of service will be automatically delivered. Opponents say that new bandwidth hungry applications will be created and quality will degrade.

#### *IntServ*

The essence of IntServ is to reserve resources for each individual flow so that the service quality can be guaranteed. Before starting the session, an application must specify its requirements and an admission control routine decides whether the request for resources can be granted. This approach is similar to the traditional telephone-switching infrastructure where resources are reserved for each call. One problem is the increased burden on the routers, which need to know and store the state of many flows at the same time. Moreover, all of them along the connection between the two end-points must be enabled to support protocols and procedures like RSVP, admission control and packet scheduling.

#### *DiffServ*

DiffServ divides traffic into different classes and gives them differentiated treatment. To distinguish the classes, the *Type of Services* field is used in the IPv4 header. Using different classification, policing, shaping and scheduling rules, several Classes of Services can be provided: this decision is in charge of the Internet Service Provider. There are only a limited number of service classes, which means that the routers have to store data proportional to the number of classes and not depending of the flow. DiffServ is therefore more scalable than IntServ and easier to implement. There is still an unsolved problem when high priority traffic concentrates on one router: the bandwidth can be so saturated that it adversely affects performance.

#### *Traffic engineering and redirection*

Traffic engineering is an iterative process of network planning and optimization. Network planning improves the architecture of a network in a systematic way so that the network is robust, adaptive and easy to operate. Network optimization means controlling the mapping and distribution of traffic over the existing network infrastructure in order to avoid and/or relieve congestion, to assure satisfactory service delivery.

Traffic redirection is based on content replication or caching. It starts analyzing the location of the requesting computer and defines from where the resource should be delivered, choosing the nearest or the less congested path to any of the worldwide distributed servers. Leading companies like Akamai or Digital Island are gaining market shares especially for streaming video content.

**The Videonet solution**

Videonet, a Telecom Italia product, is the first IP-based commercial videoconferencing system supporting Class of Service. It was launched in 2000. The architecture aims to provide the Premium Class of Service with reliable, low delay and low jitter service. So far it only supports two-way videoconferencing, but future enhancements are planned.

The approach to provide QoS is essentially the DiffServ one, with an additional control system to ensure that the total Premium traffic does not surpass the total network capacity. Each customer subscribes Service Level Agreements with Telecom Italia, specifying a desired committed bit rate, whose deliver is granted from the network even though there is congestion. If the customer exceeds the committed rate, the excess traffic is dropped when necessary. The Type of Service field of the Premium traffic is set to a particular value, so that the network can recognize it: this process is called traffic marking or *colouring*. So far with Videonet only the videoconferencing traffic can be *coloured* for a H.323 session.

Policing is used to determine whether the traffic from a particular customer conforms to the traffic profile specified in the Service Level Agreement: an action is accordingly performed. In this context the edge router controls that only the videoconferencing traffic is marked as Premium traffic.

Custom Queuing allows sharing the network resources among applications with a specific minimum bandwidth requirement. It handles the traffic by assigning a specified amount of queue space to each class of packets. In this phase of the Videonet implementation there are only two queues, for basic and for Premium traffic. Associated with each output queue there is a configurable byte count, which specifies how many bytes of data should be delivered from the current queue before moving on to the next queue. The values are set so that 90% of the bandwidth is reserved to Premium traffic. The router services the queues by cycling through them in a round-robin fashion, sending the portion of allocated bandwidth for each queue before moving to the next queue. If one queue is empty, the router will send packets from the next queue that has packets ready to send. The customer can exceed the committed rate. If there is congestion, it is notified to the router, that reduces the packets throughput.

The cost plan of Videonet has two proposals: flat rate or pay-per-traffic. In case of heavy use, the first proposal may be cheaper, but at the beginning the pay-per-traffic solution is more convenient. The savings, comparing to ISDN solutions, can be very high when dealing with distant locations. There are plans to use only HDSL connections instead of leased lines, which may lead to the cancellation of the flat rate proposal.

**The Videonet test site implementation**

A 512 Kbps leased line has been installed in both premises, the “Campus Bio-Medico” University Hospital of Rome and the Department of New Technology of ELIS Consortium. The lines are connected to the Telecom Italia Interbusiness public network and handle normal Internet traffic. Two Cisco 2600 routers (one for the routing and one as a gatekeeper) manage the normal and the Premium services. Ten HP personal computers equipped with Aethra 9200 boards for videoconferencing are connected to the two Local Area Networks. They run different Windows operating systems, namely 95, 98, NT and, when the Aethra drivers will be released, also Windows 2000, in order to have a complete test on all typical platforms. A webcam and a headset with microphone provide the audiovisual input.

A videoconference can now be set at speeds of 128, 256 and 384 Kbps. We have plans to implement a faster connection on a HDSL line, in order to test 768 Kbps. This level of speed should give a superior quality compared to the typical high-end 384 Kbps ISDN videoconferences.

Physicians tend to have a subjective feeling of the quality of what they see on a monitor. They may even judge better (because of higher contrast) lossy compressed images (like JPEG for still images and MPEG for movies) than the original ones: clearly, when zooming on details, the truth comes out. The compression level is an important factor: if it is low, there are studies demonstrating that the diagnostic analysis can be performed with almost no risks [6].

In order to evaluate the difference between a QoS videoconferencing session and another without quality control, we used the ITU-T Recommendation P-800 [7]. We submitted the *conversation-opinion test* to medical and engineering students who were at their first videoconferencing experience. The tasks for every couple of testers, one at each side, were: name-guessing, story comparison, building an object, picture-comparison. We ensured that there were proper lighting, low background noise and proper dressing (no bright colours, no stripes), in order to facilitate the encoding. Short duration and fully written instructions provided the lowest bias for the tests. At the end of each task, each student answered four questions regarding the perceived level of audio, video, audiovisual experience and easiness of interrupting the other person while speaking (this question measured the quality of the interaction).

We are aware that the purpose of a video connection is always a discriminating variable: we would not judge with the same parameters a talk between two physicians and a live interaction with the operating theatre for remote consultation or aid in the intervention. Therefore our results are only the first step to a complete assessment of the Videonet system in a hospital environment.

## Results

For the perceived QoS tests, using ITU-T P.800, we added a non-Premium traffic (noise-traffic) of 390 Kbps to a 128 Kbps videoconferencing session in order to completely saturate the available bandwidth. Table 1 shows the statistics on the answers of the N students involved in each test. We chose not to submit the same test (with and without QoS) to the same person because we think that the first one might influence the second identical test.

*Table 1 – Scores for subjective quality tests at 128 Kbps mean value ± standard deviation*

| **Question about** | **Scale** | **N** | **With QoS** | **N** | **Without QoS** | **P** |
|---|---|---|---|---|---|---|
| Audio | 1-5 | 15 | 4.13 ± 0.83 | 14 | 3.29 ± 0.73 | 0.01 |
| Video | 1-5 | 15 | 3.60 ± 0.83 | 14 | 3.07 ± 0.62 | 0.06 |
| Audiovisual | 1-5 | 15 | 3.73 ± 0.70 | 14 | 3.21 ± 0.70 | 0.06 |
| Interaction | 1-4 | 14 | 3.42 ± 0.85 | 14 | 3.07 ± 0.92 | 0.30 |

The P value indicates the probability of the null hypothesis. In the evaluation of the audio quality during the session there is a statistically significant difference (P<5%) between the two mean values, which gives a good chance of having detected a perceived improvement with QoS enabled. Video and audiovisual questions are on the border of statistical significance, while interaction does not seem to be better in the QoS environment. We think that the reason may be in the Custom Queuing algorithm. If the Premium traffic is within the bandwidth limit (128 out of 512), the algorithm delivers all the *coloured* packets even though there is noise-traffic that exceeds that limit (390+128>512): only non-Premium packets are therefore discarded. On the contrary, if QoS is not enabled, also some of the packets of the videoconferencing session are discarded. This scenario explains the good audio quality recorded with QoS enabled: all packets are delivered and sound quality is preserved. On the other hand, a good interaction depends primarily on a low Round Trip Time, that is the absence of long delays. With Custom Queuing, unfortunately, the RTT of *coloured* packets can be high if the total traffic (Premium plus non-Premium) is congested, because of the increased length of the queues: it takes time to handle them in order to decide which to serve first. A RTT over 400 ms is considered unacceptable for any audio or video interactive session and we reached it in some of the tests.

This is an important issue because it seems to show that videoconferencing, which requires real time interaction, is not the most suitable application in a QoS environment based mainly on Custom Queuing.

Similar tests with 256 Kbps videoconferencing did not show a significant difference with and without QoS. We discovered soon after these tests that there was a failure in an ATM interface in the routing path, which caused a lowering of the maximum bandwidth from 512 Kbps to around 300 Kbps. For a similar reason, tests at 384 Kbps failed. Therefore we discarded all these tests planning to repeat them after solving most of the other problems.

Other tests compared the subjective quality of a 128 Kbps IP (H.323) session with the ISDN (H.320) videoconferencing at equivalent speed. The results always showed a lower quality for the IP session, with and without QoS. The H.323 Aethra implementation uses only the G.711 audio codec, which requires 64 Kbps while the H.320 can use G.722 (48 to 64 Kbps) and G.728, which requires only 16 Kbps. The use of G.728 leaves more bandwidth to video transmission, while retaining a perfect speech comprehension. As the Aethra SV9200 board does not support more than 128 Kbps for ISDN we could not compare qualities at 256 or 384.

A failure in the Videonet system is the impossibility of starting a videoconferencing session if the bandwidth is already saturated by normal traffic. The reason may be that the signaling traffic of the H.323 session is not *coloured*, therefore it does not overcome the non-Premium traffic and the *coloured* session never starts.

Application sharing suffers from the problem of long RTT when traffic is congested. Moreover, it has been implemented using Microsoft Netmeeting 2.11 which runs worse than version 3.01. In addition, Netmeeting generates a software conflict with the Aethra software if it starts before the videoconferencing session.

Our tests with different Windows operating systems showed that the driver implementations of the videoconferencing board have different performances. The NT version has problems in the RTP/RTCP stack. Most of our tests were therefore on the Windows 98 machines.

Interoperability among videoconferencing H.323 boards of different makers is an open issue because some of them are not compatible. The provider’s suggestion is to have all boards of the same kind, but this is not an optimal scenario: it can be applied for calls within the same institution but it would fail when calling external parties.

Another problem arises when the client computers are behind a proxy, which operates Network Address Translation (NAT). This is quite frequent in a hospital because of security reasons, or for limited availability of IP numbers (with a proxy, only one external public IP number is needed even though there are hundreds of computers connected), or for optimizing caching of web content. So far Videonet, like other videoconferencing systems, is not able to connect through a proxy. We believe that it is necessary to overcome this limitation.

We tested the system also on the internal LAN, that is without using the QoS routing on the public Internet. Some limitations have been noticed which should be solved by properly configuring the gatekeeper. Being the LAN a fast Ethernet (100 Mbps), the behaviour is completely different from the external connections and requires a different analysis. When just a few users are involved, bandwidth is never saturated. A white paper by the videoconferencing leader CUseeMe [8] reports that congestion shows up with more than 50 concurrent video users.

## Discussion

Quality of Service over IP applied to videoconferencing is surely a great advance in the field. As long as it is widely enabled, it can give a boost to telemedicine because of the increased facility to set up a video connection in any Internet connected point. Even though some experiences [9] show optimistic views of the existing technology, we are aware that the transition from enthusiasm in experimentation to frustration in daily work is almost the normal situation nowadays. Moreover, the open technical issues of Videonet show that there is still work to be done to achieve a quality comparable to ISDN. The main advantage of Videonet so far is that a videoconferencing session could be set from any place connected to the network, which is normally ubiquitous in a hospital.

We believe that it is necessary not to limit the use of Videonet only to H.323, in order to be able to provide a wider range of services with guaranteed quality. We are going to work with Telecom Italia Lab, the research center which developed Videonet, for enabling the same network to accept streaming video applications. This would give an end user the possibility of video-on-demand and live one-to-many transmission at a high and constant quality. Video based training would benefit of this scenario. Cisco Systems has supported us with a complete Cisco IP/TV infrastructure that could allow us to deliver live webcasts from the operating theatre at high and guaranteed quality to many users in different locations.

## Conclusion

The project is going on and widening its initial goals. Multiconferencing and multicasting, which now are not addressed by the Videonet solution, will be further steps to improve the QoS framework after solving the main problems detailed in this paper.

We plan to work with Media Xchange Manager, a new product by VCON aimed at simplifying the management of many videoconferencing stations, to make them as simple as a normal telephone. It is a sort of PBX for IP traffic, which hides from the user all the complexity of connecting procedures. We believe that the combination of Quality of Service and easiness of call management can be the *killer application* of videoconferencing in the next years, allowing more medical environments to make use of it in daily work.

Further information and ongoing work about our project is available at http://research.unicampus.it/videonet.

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