SISTEMI DI ELABORAZIONE

MULTIMEDIAL STREAMING TOWARDS PDA

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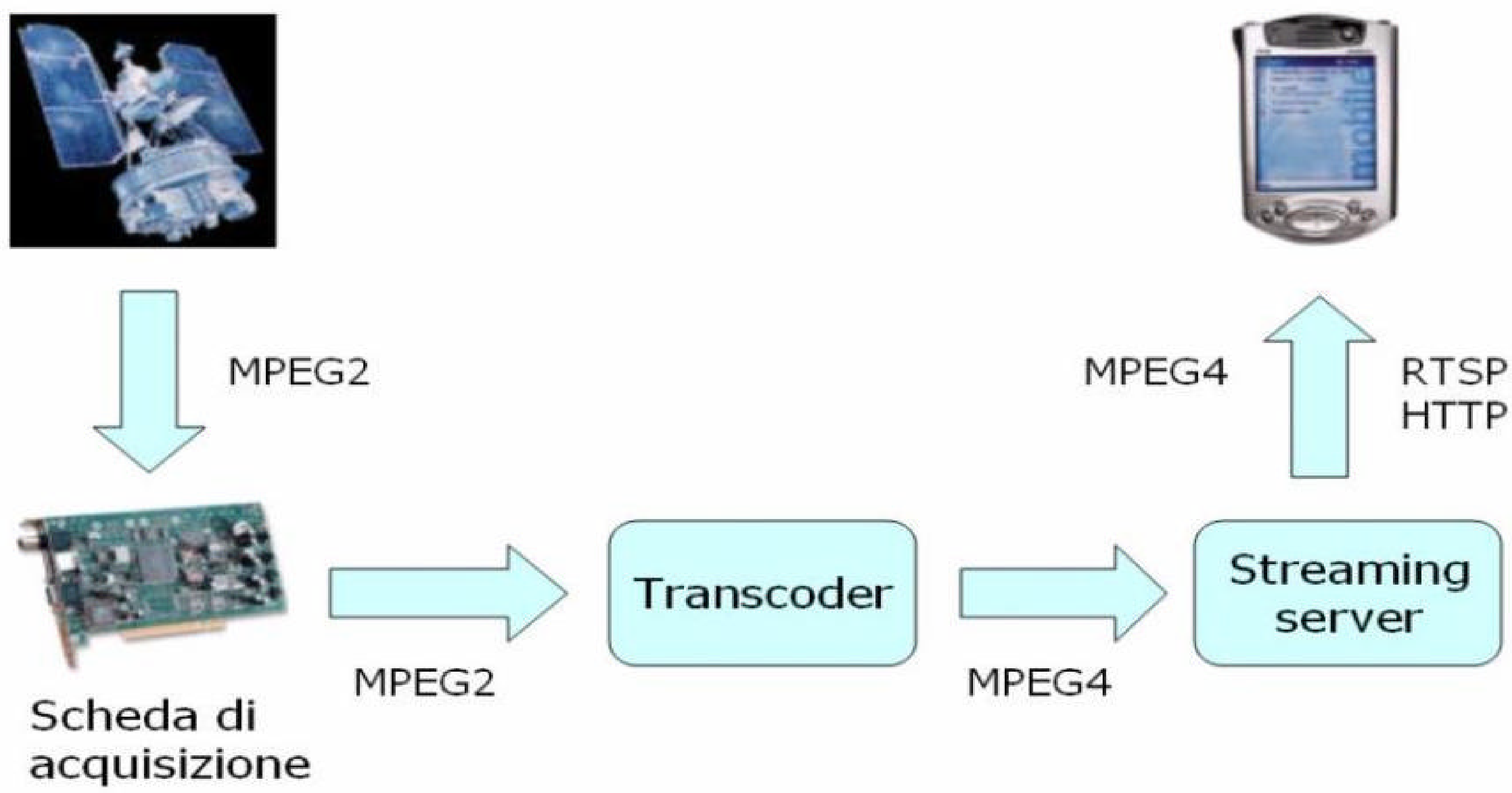
PROJECT OBJECTIVE

The final purpose consists of a visualization of a multimedial audio-video flow from a satellite on a PDA display. The flow, coming from a satellite trough a MPEG2 code and acquired with a specified card, before being sent via wireless to the client, it needs a further transcodification in a format that allows an efficient streaming and a good quality too. It has token the MPEG4 format.

PRINCIPAL SCHEME



THE MPEG FORMAT

MPEG is the acronym of Moving Pictures Expert Group and it is an ISO standard that settles the digital video and audio standards. The aim of the group that created it was to create a writing format for audio and video in real time from a CD, but it continued to include in this project the transmission from a satellite, from networks and from DVD (Digital Versatile Disk) too. We got, in that way, standard with different profiles such MPEG1, MPEG2 and MPEG4.

THE MPEG FORMAT

Format	Use	Comment
MPEG1	Video CD	First revolutionary video format MPEG. VHS quality.
MPEG2	DVD Satellite	MPEG standard for high resolution. Bit rate very high.
MPEG4	WWW Video CD	New version with a bigger compression. More familiarity to the streaming and the interactivity.

THE HTTP PROTOCOL

The HTTP protocol, based on TCP, explains an interaction client-server method that optimize the reliability of communications. It refers to the *application* level of OSI standard and represents a protocol stateless and light.

Stateless means that the server doesn’t have memory of connections done so it uses all the connections at the same way. Light means that the client connects to the server only for the time necessary to broadcast the resource.

FROM TCP TO UDP

Nevertheless for the multimedial streaming requirements, the critical parameter, more than reliability, is the transfer speed of data flow.

For this reason, the most of the video servers don't use HTTP and TCP, but utilize UDP (User Datagram Protocol) as base to manage the continuous flow of data packages.

THE RTP PROTOCOL

The RTP protocol (Real Time Protocol), which is based on UDP, doesn’t guarantee the supply of all packages, but it only offers a flow as fast as possible, supplying working mechanisms of time and to number the sequences to ensure that all the coming data packages could be placed in the correct order.

RTP forms an efficient framework on which realize multimedial applications.

THE RTSP PROTOCOL

The RTSP (Real Time Streaming Protocol) is a level applicative protocol which leans to the Real Time Protocol to supply services of multimedial streaming.

We could considerate the RTSP protocol as the TV remote control and the RTP protocol as the responsible for transport from the remote control to TV.

THE RTSP PROTOCOL

The RTSP is an applicative protocol, so it offers a wide range of functions comparable to a video taper recorder (play, pause, rewind, FF …), while RTP occupies itself to manage the data flow between the media service (who has the data) and the client/s connected to it.

THE SYSTEM

The acquisition card from a satellite and the required softwares have been installed on a system with the following characteristics:

- CPU: *AMD Duron 800 MHz*
- RAM: *256 MB PC133*
- Hard Disk: *Samsung 8 GB 5400 rpm*
- Operative System: *Linux RedHat 8.0*

THE DVB CARD

- To acquire the MPEG2 flow from a satellite has been installed a DVB (Digital Video Broadcasting) card. This card is produced by *Hauppauge*.

- The card integrates the function of hardware decoding from MPEG2 with the analogic signal.

THE DVB CARD

- The drivers for installation can be found at URL *linuxtv.org*, useful reference for all that regard the streaming in the Linux environment.
- Together with the drivers, some utilities are given which allow to change channel or display the output on the monitor.

MPEG4 ENCODING: MPEG4IP

MPEG4IP represents a range of open source applications which allow to manage an MPEG4 flow, from the encoding to the visualization.

MP4LIVE is the software that permits the encoding of an audio-video flow in the MPEG4 format, offering the possibility to set the parameters to encoding the flow (resolution, frame rate and bit rate for video and sampling rate e bit rate for audio).

MPEG4 ENCODING: MPEG4IP

Since the moment MP4LIVE doesn't accept in input the MPEG2 digital flow coming from the satellite, it has been very useful the hardware decoding function of MPEG2 flow in analogical supply from DVB card.

The encoding result of MP4LIVE could be saved on a file or instead used by the streaming server to realize a live transmission.

THE SDP FORMAT

SDP (Session Description Protocol) is part of the IETF standard format (Internet Engineering Task Force) to describe RTP audio and video flows.

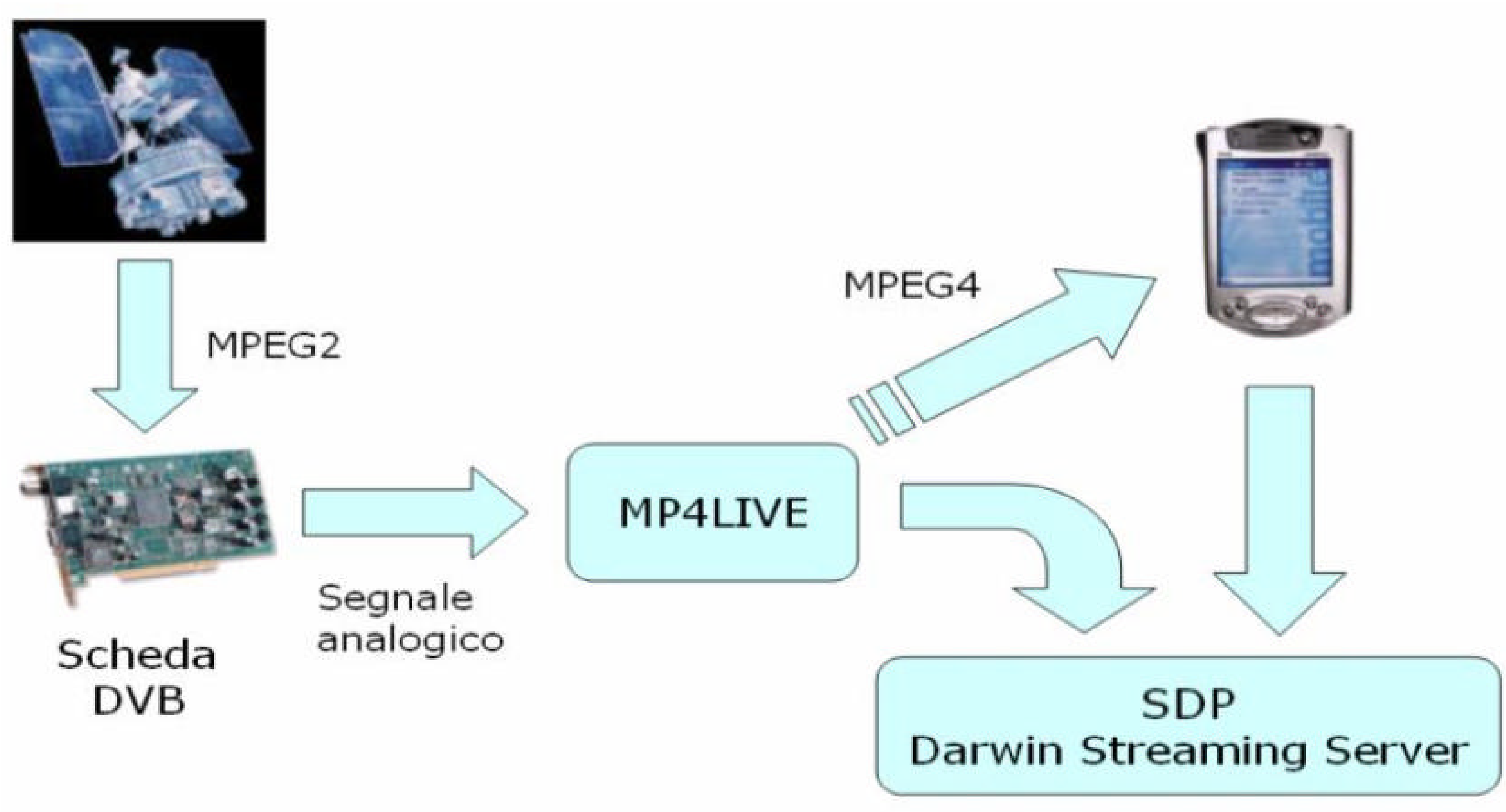
A SDP file contains informations about the format, the source and timing of a multimedial flow.

THE SDP FORMAT

The SDP make possible the streaming of multimedial contents in real-time:

- The encoding software, in this particular case MP4LIVE, creates a SDP file to be put in the work directory of the streaming server;
- The client, through every player which support RTSP or HTTP, asks to the server the SDP file;
- The server provides to connect the client with the multimedial flow source represented by the encoding program.

PRINCIPAL SCHEME



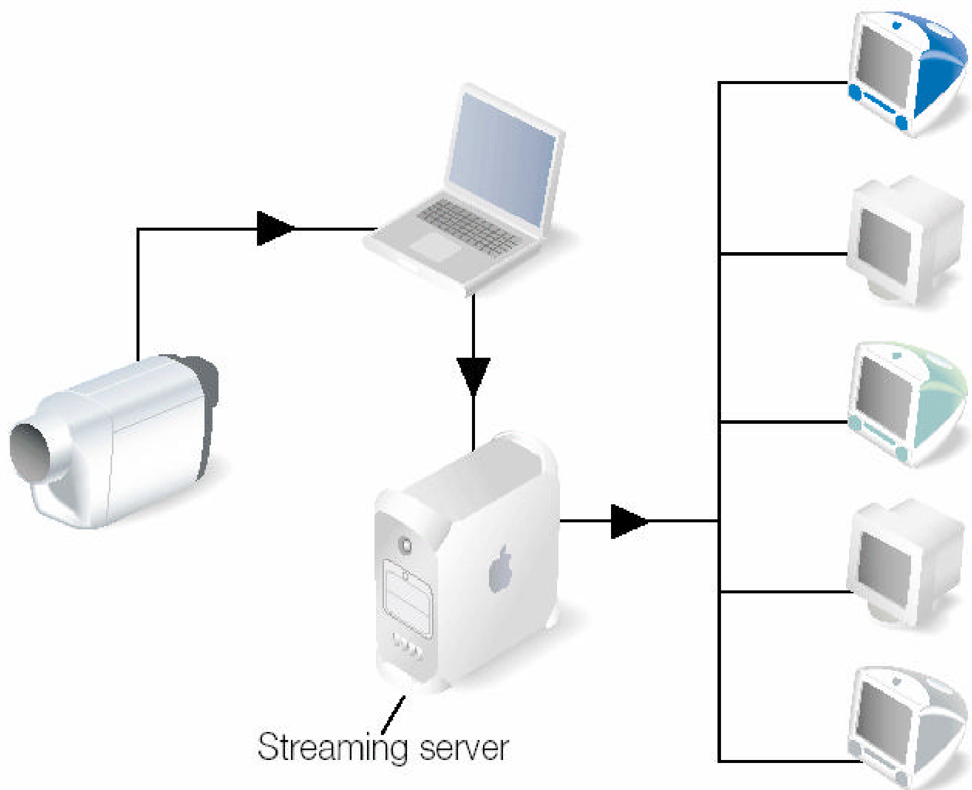
THE STREAMING SERVER

The streaming server has got a general function to dispose multimedial contents to be used by the clients.

Using, as said already, the SDP description, the server can provide to clients a real-time flow.

DARWIN STREAMING SERVER

DSS is an open source produced by Apple and token from QTSS (QuickTime Streaming Server).



DARWIN STREAMING SERVER

- It can send multimedial flows in every format (MPEG4 and MP3 as well);
- For the transmission it supports RTSP and HTTP;
- It allows to set playlists in order to create radio stations or TV on the Web;
- It has also security mechanisms through the SSL protocol (Secure Sockets Layer).

THE PLAYER CHOICE

At the moment readers don't exist for *PocketPC* compatible with the streaming in the MPEG4 format.

The best product available at the moment is *RealOne* by *Real Software*, the only one that support the RTSP protocol. It doesn't support the streaming in MPEG4 either, but, possibly in the near future, it'll be created a specific plugin by the company.

THE PLAYER CHOICE

Other softwares exist to read the MPEG4 files, such as *PocketDivX* and *PictPocket Cinema*, but they are version even not completed and they don't offer the possibility to receive a flow from a remote server either.

It has been nevertheless possible to realize a streaming from Darwin to PDA, but only in MPEG1 format through the player *PocketTV*.

TESTING

Finally, we tested the transmission both of pre-recorded contents than the flow in real-time.

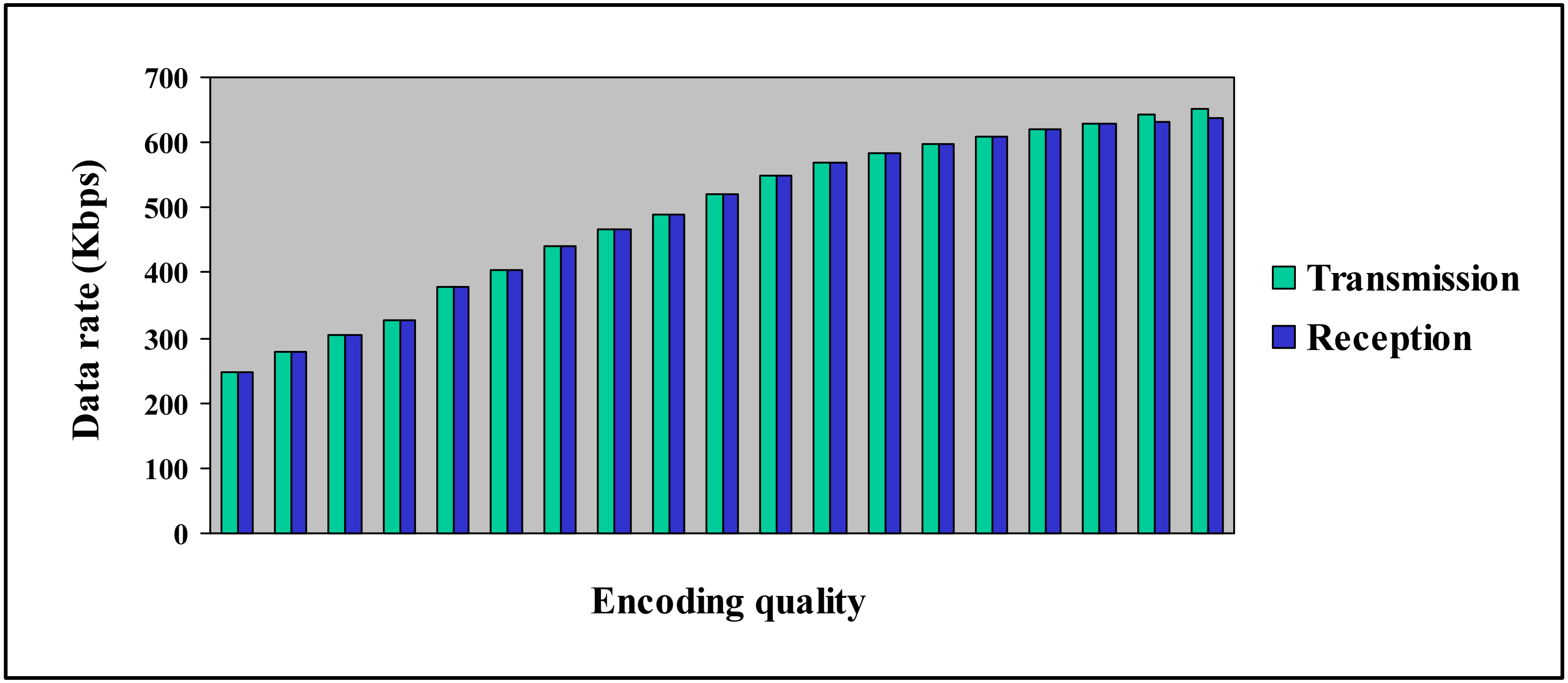
As client it has been used a notebook connected on the LAN with the server via wireless. The player chosen has been *QuickTime Player*, through which it has been possible to value the data rate received and the percentage of lost packets.

TESTING

We did several tests changing the encoding parameters of the flow as in the table below, keeping constant the dimensions as 320×240 pixels.

		Min value	Max value
Video	**Frame rate**	12 fps	24 fps
	Bit rate	225 Kbps	4000 Kbps
Audio	**Sampling**	8 KHz	48 KHz
	Bit rate	32 Kbps	320 Kbps

TESTING


700
600
500
400
300
200
100
0
Data rate (Kbps)
Encoding quality
Transmission
Reception

TESTING

We used a network with a very large band (10 Mbps), so the performances obtained resulted satisfactory in all the situation tested. We had only a small loss of packets (~ 3%) when we adopted the best quality of transaction possible.

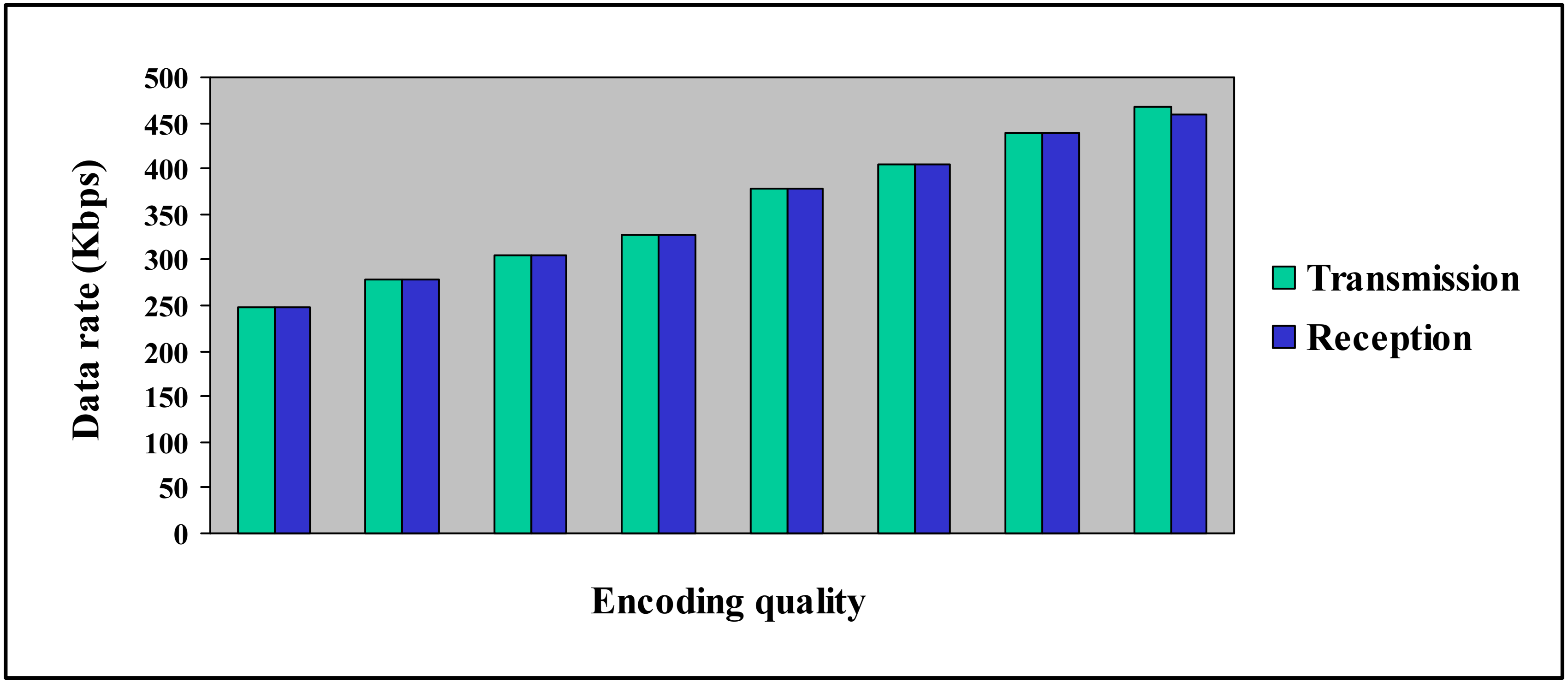
We tested also if this loss has due to an overload of the network or a lack of server's performances.

TESTING

In this case we've done a lot of tests visualizing high quality multimedial contents directly from Internet.

Since the results have been observed they've always been very good, we deduced that the cause of the packets loss are the server's performances.

TESTING LIVE



TESTING LIVE

In live modality we obtained acceptable performances till the following encoding configuration:

Video	**Frame rate**	18 fps
	Bit rate	1425 Kbps
Audio	**Sampling**	44.1 KHz
	Bit rate	128 Kbps

In these conditions it's although guarantee a good quality both audio and video.

TESTING LIVE

If you decide to adopt more expensive encoding configurations, the server's calculation capacities are not enough (the server is responsible both the encoding than the transmission) and the high desyncronization of audio and video flows (or in some extreme cases, the absolute absence of one at least) causes an absolutely unacceptable quality.

TESTING

To obtain a satisfactory quality it's also necessary report the encoding parameters with the type of program we want to transmit.

In fact, with equals encoding parameters, with many framing changes (for example sport events) we got a bad quality results.

FUTURE DEVELOPEMENTS

In the short-middle period it will be ready some PDA players adapt for the streaming in MPEG4 format, making this service effectively realizable.

At this point it'll be interesting to test the streaming server's performances if it needs to serve at the same time a lot of PDA clients or with a strong traffic on the network.

FUTURE DEVELOPEMENTS

In this case it's important to underline the RTSP protocol, used in the testing phase, doesn't guarantee any type of quality of service.

To really dispose of a good multimedial streaming service it'll be necessary to adopt instruments which are able to guarantee mechanism to reserve resources.