Beyond 3G: A Multi-Services Broadband Wireless Network with Bandwidth Optimisation

George Gardikis♦, Evangelos Pallis♦, Anastasios Kourtis♦

♦ National Technical University of Athens, Mobile Radiocommunications Lab., 9 Iroon Polytechneiou, 15773 Zografou, Athens, Greece.
♦ Institute of Informatics and Telecommunications, NCSR "Demokritos" 153 10 Ag.Paraskevi, Athens, Greece.
e-mails: {gardikis; pallis; kourtis}@iit.demokritos.gr

ABSTRACT

The main goal towards which nearly all research activities in the field of digital wireless communications are oriented is the development of a global wireless access network, capable of offering real-time bandwidth-demanding multimedia applications and enabling for anytime/anywhere access. This paper proposes a wireless access network which is a result of the convergence of telecommunications and digital broadcasting, two sectors in which Europe is a global pioneer. The proposed network, whose implementation will be undertaken in the frame of the EU-funded MAMBO[1] project, can offer digital television and fast Internet access to both mobile and stationary users, while achieving bandwidth optimization through perceptive QoS estimation and transcoding algorithms. DVB-T technology is used for broadband downlink, while the uplink is based on present cellular platforms (GSM for mobile and LMDS for residential end-users).

I. INTRODUCTION

Telecommunications and digital broadcasting have been recognized as two sectors where Europe has acquired a significant technological and commercial leadership. Until recently, both industries have followed parallel paths, with conventional telecommunications operators being mainly interested in the technology used for information transport and delivery and traditional broadcasters being interested in content. Nowadays despite the intrinsic technological differences between these two sectors, the question is being openly asked as to the likelihood of their convergence, not only at technological level, but also at service level. Broadcasters are progressively introducing an element of interactivity in their traditional point-to-multipoint channels and telecommunications operators are confronted with traffic streams which are inherently asymmetric.

Tightly related to the above developments are the phenomenal growth in mobile communications and the development of Internet based services. Broadcasters that have started to use internet as a delivery mechanism are contemplating the use mobile communications systems such as GSM (Global System for Mobile) and UMTS (Universal Mobile Telecommunications System) to enhance their multimedia content and mobile operators are seeking to provide user access to such content and in so doing provide a whole new range of value added services.

The forthcoming 3G technologies can offer bi-directional packet-switched traffic at rates up to 384 kbps (WCDMA approach) for wide-area coverage, a bit rate sufficient for high-speed Web access, videotelephony or streaming small sized MPEG (Motion Pictures Experts Group) video clips. Third-generation platforms cannot however handle the rates needed for high quality digital video broadcasting, something that the emerging DVB-T (Digital Video Brodcasting – Terrestrial) technologies are promising.

In the context of the European Community's 4th Framework Programme of Research and Development the area of Interactive Broadcasting was the subject of innovative work by a number of projects, and major contributions were made to both the development of mobile and digital broadcasting technologies. The IST (Information Society Technologies) programme recently launched in the context of the 5th Framework Programme provides further opportunities for addressing the technological and service dimensions of Interactive Broadcasting.

Among the projects funded by the EU in the IST framework is MAMBO (IST-2000-26298) whose goal is to develop the network illustrated in this paper. The key objective of the MAMBO project is to develop, implement and assess an universal, open and scalable platform that manage the distribution of high quality interactive multimedia DVB / IP services through a terrestrial access network, to mobile and residential end-users by an optimal allocation of the allowable bandwidth. A distributed feed back loop bandwidth optimisation mechanism will be implemented, which is able to adapt, in real time, the bit rate of each service according to the available bandwidth and the complexity of the service, without degrading the service quality.

The network proposed in this paper combines the DVB-T concept with existing cellular technologies such as GSM and LMDS to offer digital television services through terrestrial networks to mobile or stationary users along with high-speed data services. By using DVB-T for the downlink and GSM/LMDS for the uplink, an asymmetrical access network is created which takes advantage of the main characteristic of platforms providing multimedia content: The amount of data retrieved by the end user is huge compared to that which is sent back to the network, mostly in the form of requests or acknowledgements. The DVB-T downstream comprises a CBR multi-megabit MPEG bouquet able to carry DTV programs along with IP data.

There is also one critical issue to be taken in mind when dealing with provision of multimedia services: the quality of sound or picture presented to the end user –assuming a high-quality source and an error-free environment- is directly proportional to the bitrate used in the encoding process [1]. The objective quality perceived by the non-expert user is referred to as the "Percepted Quality of Service" (PQoS), measured with purely subjective criteria, as opposed to the Network QoS which relies on objective measurable parameters (throughput, BER etc.). By altering the encoder rate, the network administrator can affect the picture or sound quality, and achieve the "golden section" between quality of service and bandwidth availability. The innovation introduced by the proposed network is the automatic bandwidth optimization achieved through a process of transcoding the incoming media streams and estimating the PQoS on the user side. A high-rate multiplex is thus created, dynamically adjusted to satisfy constantly changing parameters and requirements.

II. OVERALL SYSTEM CONFIGURATION

The proposed architecture is the solution for service providers and mobile operators who demand a flexible and cost-effective method for managing compressed digital services while maximising bandwidth capacity but maintaining the service quality. At the service provider site, the operator is able to select specific services of interest, from a large number of TV and IP services, locally generated or arriving at its premises via satellite, terrestrial or cable networks. Then, the operator defines a priority service list for the selected TV programmes, as an input to the transcoder and the statistical remultiplexer. The system automatically allocates the bandwidth to every service, changes the bit rate of the service, if necessary, and realises a statistical optimised multiplex, based on the feedback inputs from the available channel bandwidth, the IP traffic and the estimation of the quality of service. This solution, available in real time, enables operators to statistically remultiplex services from a variety of sources and create a new, customised statistical

multiplex while optimising bandwidth for delivery of additional tiered services.

The overall system configuration is illustrated in Fig.1. The use of the DVB-T platform for the downlink provides an extended coverage area (macrocell) within which the client terminals, either mobile or stationary, can operate. The return channel is provided by a cellular platform (GSM for mobile and LMDS for residential users) organized in microcells inside the DVB coverage area. The microcell nodes feed the return data (requests, acknowledgements, QoS parameters) back into the system kernel through microwave links.

The kernel of the platform, installed at service provider level, consists of the stream transcoder and the bandwidth allocation management system along with the statistical remultiplexer. The architecture of the kernel along with the feed back loop optimization mechanism is depicted in Fig.2. The transcoder receives all kind of digital streams (Video/Audio/IP data) and adapts their bit rate in compliance to commands generated from the bandwidth allocation management system. The outputs of the transcoder are fed into the statistical remultiplexer which
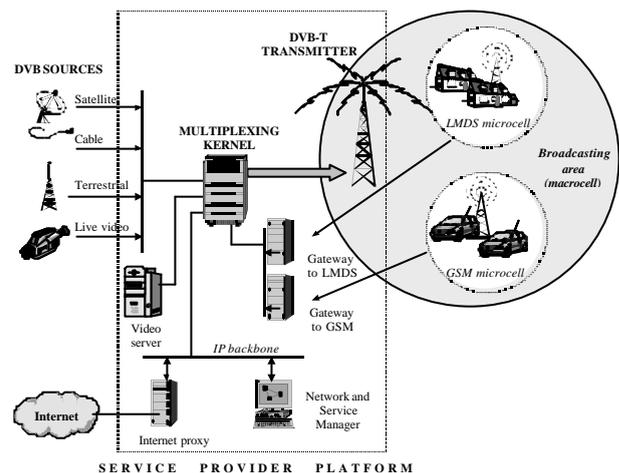

Figure 1: General System Configuration

forms the CBR DVB bouquet. The final multiplex is the input to the COFDM (Coded Orthogonal Frequency Multiplexing) modulator and power amplifier and is transmitted in the UHF band according to the DVB-T standard (ETS 300 744).

At the receiver side, the incoming broadband data are demultiplexed, decrypted (if necessary) and presented to the user. An intelligent interface module extracts the Perceptive QoS parameters for the media stream being presented and sends the QoS data back to the service provider, whose responsibility is to decide for the adaptation of the bit rate of the media presented to the user.

III. DETAILED NETWORK DESCRIPTION

A. The statistical multiplexer

It is the "heart" of the whole network. It allows the network administrator to create a multiplex of his choice, consisting of television channels, VOD and N-VOD services, and IP data. It combines all these inputs to form an MPEG-2 Transport Stream suitable for transmission in accordance to the DVB-T standard and performs dynamic allocation of the available bandwidth by adapting continuously to the bit rate variations of the incoming streams while sustaining the rate of the multiplex constant. By using statistical
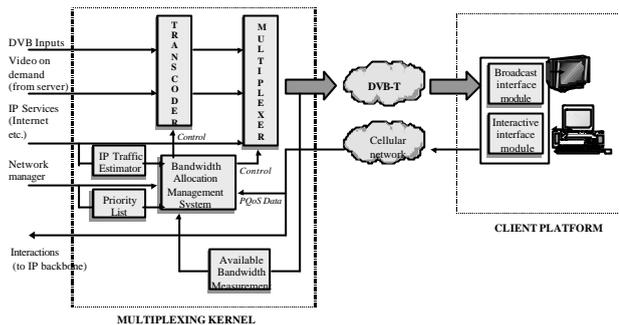

Figure 2: The bandwidth optimisation mechanism

instead of static time-division multiplexing, it does not waste bandwidth, as it happens when static resources are allocated to variable bit-rate channels. Moreover, it must be fully programmable in order to adjust the throughput of non-time-critical services (e.g. Internet access), depending on the available bandwidth. The MPEG-2 standard is used for the representation of digital video and audio, while the encapsulation of IP data in MPEG-2 Transport Stream packets follows the DVB-SI recommendation.

The multiplexer receives commands from the network administrator and the bandwidth allocation management system in order to select the input streams and to effectively distribute the available bandwidth.

B. The stream transcoder

Each transcoder module accepts an MPEG-2 digital video stream as input and produces a stream of a lower bit-rate than the original, degrading the picture and sound quality, but maintaining the content of the sequence and the video standard (picture rate, aspect ratio, colour format etc.). The ideal transcoder should introduce no more additional degradation of image quality than necessary and only an imperceptible delay, while it would be designed aiming at a low-cost implementation.

The simplest solution for digital video transcoder is to cascade one standalone encoder and one standalone decoder, configured to operate at a lower bit rate. The compressed video is uncompressed to a raw digital

stream and then re-compressed. However, this design introduces additional compression impairments to the picture, even if the bit rate was not to change. In other words, it degrades the picture quality more than necessary. Moreover, the work done by the encoder of the previous generation is of no use, and the new encoder must perform again all complex operation such as motion compensation etc. A "smart" transcoder performs mathematical functions and truncation within the existing MPEG-2 program and re-multiplexes video and audio elementary streams to produce a synchronised Transport Stream.

The stream transcoder accepts commands from the bandwidth allocation management system and adapts the bit rates of the incoming streams to satisfy the requirements of the network. If the quality of a video program is to be degraded, either because of bandwidth shortage or more-than-sufficient perceptive quality on the client side, the transcoder reduces the encoding bitrate to match the new settings.

C. The bandwidth allocation management system

It is a feed back loop bandwidth optimisation mechanism which controls how the bit-rates of each incoming service are to be altered, either through transcoding for video and audio channels, or by plain throughput constraining for IP data services. In the first case, the command for bitrate adaptation is sent to the transcoding engine, while in the second a simple request directly to the statistical multiplexer is sufficient. The System must incorporate an efficient and, at the same time, fair algorithm which at frequent intervals decides how much bandwidth is to be allocated to each service. The following parameters should be taken in mind for this decision:

- The bandwidth of the final multiplex and the maximum bit rate (defined by the multiplexer and/or the transmitter) whose value should never be exceeded.
- The number of signals that are fed into the multiplexer. Again, sources coming from different content providers can be unequally treated, upon agreement with the network administrator. In both of the above cases, the administrator forms a service priority list for the incoming streams. The bandwidth allocation module has to confirm to the priorities described in this list.
- The Perceptual QoS data for each incoming stream, as this is evaluated on the client side and is provided by the return channel. For example, movies with stationary scenes can be transcoded with lower bit-rate without significant degradation of their quality. The bandwidth gained can be dynamically allocated to more demanding streams, like sports programs with rapidly changing scenes.

D. Perceived QoS Management System

The P-QoS Management System measures the perceptual quality of the services provided to the end users. The results are fed through the return channel into the bandwidth allocation management system so that the gap between the measured QoS and the required QoS level is minimized.

The main issue in designing the P-QoS evaluator is the characteristics of the image which define how annoying the degradation of the quality is to the average user.

Performing Perceived QoS measurements on a bit-stream requires to associate data related to QoS to the bit-stream. In this process, certain video picture characteristics are measured and the results are inserted in the bitstream to be transmitted as ancillary data. These data characterise the video pictures at the input of the platform, before transcoding and transmission. At the end-user level, the Perceived QoS can be measured using a P-QoS measurement module similar to the one used at the transmission platform. The module further proceeds to the comparison of the actual video pictures characteristics to the original characteristics that are present in the bitstream as ancillary data, and derives the P-QoS measurement. The quality measurements are sent back to the platform through the return channel. If the P-QoS offered is significantly lower than the one which corresponds to the service, then the platform modifies (some of) the network parameters so as to minimize and possibly eliminate this deviation.

The P-QoS Management System bases its operation on a knowledge approach. It relies on a database, which contains perceptual video quality evaluations. These can be obtained by means of an extensive campaign of subjective test and objective measurement calculations.

E. The wireless access network

The wireless access network provides the connectivity between the remultiplexing platform and the end users. For the downlink, the DVB-T transmission system is chosen. It can offer data rates from 5 to 32 Mbps [2], speeds far beyond all contemporary mobile wireless systems, while the COFDM modulation technique provides high mobility, strong error correction and wide area coverage (macro-cell) for transmission within the UHF band. On the end user side, neither sophisticated antennas are required, nor line-of-sight conditions which are "sine qua non" for other types of broadband wireless networks.

For the return channel, the end-users are grouped in micro-cells. In the case of stationary users, a local multipoint distribution system (LMDS) is used. The requests and acknowledgement packets stemming from end-users are wirelessly collected by an LMDS access point, located at the micro-cell main node. From there, they are forwarded to the appropriate gateway at the service provider site, though a microwave point-to-point link. LMDS can provide high bit rate connections for each micro-cell, which is shared equally among the corresponding end-users. Additionally, end-users are permanently connected to the service provider, and thus there is no delay for establishing a connection.

In the case of mobile users, the implementation of the return channel is based on GSM technology. The most efficient GSM service that could be used for that purpose is the General Packet Radio Service (GPRS). GPRS is a GSM Phase 2+ bearer service that provides actual packet data access for mobile GSM users. It reserves radio resources only when there is need and these radio resources are shared by all mobile users in a cell. There is no time-consuming call setup procedure like in the traditional circuit-switched GSM, a feature which makes the system appropriate for the transmission of small amounts of data at frequent intervals (acknowledgements, requests, P-QoS data).

IV. CONCLUSION

It is evident that the actual trend in the market is the convergence of networking infrastructures and the integration of fixed, mobile and broadcasting technologies, in order to create a new environment, that will enable citizens to access IST services wherever they are, whenever they want. The current trend and the challenge is to provide to end-users a variety of services (TV programs, Internet, multimedia applications, etc.) through wireless infrastructures.

This new convergence situation will lead to the design and development of tools, systems and platforms handling in optimized way different types of services and obviously signals. The merging of TV, Internet and multimedia data flows impose to assign resources with consideration of the requested quality of service. New systems are then needed to manage network resources in real time. These systems must be capable to operate services in changeable usage conditions, while maintaining the highest possible level of quality. It is clear that the main advantage of using digital transmission for audio-video services has been and will be the ability to use less bandwidth and infrastructure for reduced operations expenses per program.

The network that is proposed in this paper provides a solution which is viable, realizable at reasonable costs, and, perhaps the most important, it can be implemented by integrating present technologies and commercial infrastructure.

REFERENCES

[1] E. Pallis, G. Pantos, A. Kourtis, A. A. Alexandridis, "QoS in the Provision of MPEG-2 Programmes in a Wireless Indoor Environment", International Workshop on Mobile Communications, 1999, pp. 105-111.
[2] ETS 300 744: Digital Video Broadcasting: Framing structure, channel coding and modulation for digital Terrestrial television, European Telecommunications Standard Institute, 1996, p.40.