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Executive Summary

This document (D1.1p) investigates current and emerging end-user services which are transported via access, mobile, and in-building networks. It presents the services in a number of defined groups and specifies the characteristics of each service using a common metric.

The defined groups include basic communication services, Internet-based services (including file-sharing), video-based services, on-line virtual environments, remote technical and remote health services. The used common metrics include the bit-rate, tolerable delay, jitter and packet loss as well as the mobility, required traffic priority and security.

Totally, 31 end-user services are described. The bit-rate of the services varies from a few kilobits per second (some simpler telemetric services) to 24 Gigabit per second (for immersive Ultra High Definition TV), the tolerable delay varies from tens of seconds (for e-mail) to a few tens of millisecond (for grid computing), the jitter from seconds (for e-mail) to less than a millisecond (for grid computing), the packet loss from a few percent (on-line gaming) to less than a tenth of percent (thin client). The applications also have varying requirements on the traffic priority and security and may or may not allow mobility of the end-user.

The described services also vary in their “time to market”. Some, like e-mail and Internet browsing, represent the services of today, while some others, like remote home monitoring, location based services or Ultra High Definition Video are emerging services of the near or middle-term future. Finally, services like Web 3D and robotic assistant are more futuristic. Also, some of the today services are seen to be continuously developing with constantly increasing demands like TV evolving into High Definition TV and later into Ultra High Definition TV. This also applies to constantly developing distributed systems like peer-to-peer and grid computing. The exact time frames for the introduction of emerging services are not always easy to define, especially for the services of the medium-term and distant future, as well as it is not always possible to predict exactly the evolution of the existing services.

A clear conclusion is that the network of the future will most likely have to deal with multiple data streams having widely varying characteristics and requirements. It is also clear that even the short-term emerging services will require a further evolution of the existing networks towards faster bit-rates, optimal use of resources, better Quality-of-Service and less energy consumption.

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1 Introduction

1.1 Purpose and Scope

Today, large investments are carried out inside as well as outside Europe (notably Japan, Korea, USA) to install optical fibre in green field situations as well as an upgrade of the current copper-based access networks. These investments are made by operators and sometimes local authorities in order to satisfy the growing demands from the end-users (residential or enterprise) for the services that are becoming more numerous, requiring larger bandwidth and higher interactivity as well as putting higher requirements on the quality-of-service (QoS). These extensive investments in the access infrastructure have been programmed for the next five to ten years with an expectation of increased revenues to the service providers. These revenues are synonym of delivering to the end-users (the operators' customers) the required services. The provision of a new access or mobile network or to a lesser extent a new in-building network, whose infrastructure will last for tens of years must be made in accordance with the services that will be provisioned at present or in the future. It is the goal of this document to analyse, identify and characterise those upcoming services in a way that make it possible to dimension and specify the required network infrastructures.

The current and emerging end-user services are investigated whether they are transported (and reach the end-users) via access, mobile, and in-building networks. The purpose is to identify the relevant characteristics of these services, and to describe them using a common terminology (metric) thus providing the necessary data to infer the underlying access, mobile and in-building network specifications.

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1.3 Acronyms and Abbreviations

Most frequently used acronyms in the Deliverable are listed below. Additional acronyms can be specified and used throughout the text.

3DTV	Three Dimensional Television
3G	Third Generation (of mobile phone standards)
AAC	Advanced Audio Coding
ABR	Available Bit Rate
ACK	Acknowledge (Packet)
ADSL	Asymmetric Digital Subscriber Line
AMR-WB	Adaptive Multi Rate – WideBand
ARQ	Automatic Repeat-Query
ASPO	Association for the Study of Peak Oil and Gas
AVC	Advanced Video Coding
BBA	Broad Band Access
BBC	British Broadcasting Corporation
BCH	Bose, Chaudhuri and Hocquenghem (error correcting code)
BTS	Base Transceiver Station
CBR	Constant Bit Rate
CD	Compact Disk
CEPE	Centre for energy Policy and Economics
CO	Central Office
CPU	Central processing unit
DB	Data Base
DMD	Digital Micro-mirror Device
DSL	Digital Subscriber Line
DECT	Digital Enhanced Cordless Telecommunications
DSM-CC	Digital storage media command and control
DVB	Digital Video Broadcasting
DVD	Digital Video Disc
DWDM	Dense Wavelength Division Multiplexing
ECG	Electrocardiogram
EHC	Electronic Health Card
EHG	Electrohysterogram
EPON	Ethernet Passive Optical Network
ETSI	European Telecommunications Standards Institute
EU	European Union
FCFS	First-Come, First-Served
FEC	Forward Error Correction
FER	Frame Error Rate
FPS	First Person Shooters
FTTH	Fibre to the Home
FXO	Foreign Exchange Office
FXS	Foreign Exchange Station
GDP	Gross Domestic Product
GPON	Gigabit Passive Optical Network
GPRS	General Packet Radio Service
GPS	Global Positioning System
GSN	GPRS Support Nodes
HDMI	High Definition Media Interface
HDTV	High Definition Television
HE-AAC	High Efficiency AAC
HHI	(Fraunhofer) Heinrich-Hertz-Institut
ICT	Information and Communication Technologies

IEA	International Energy Agency
IGMP	Internet Group Management Protocol
IMAX	Image Maximum (film format)
IP	Internet Protocol
IPSOS	No meaning -- Poling Institute
IPTV	Internet Protocol Television
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ITU	International Telecommunication Union
KPN	Koninklijke KPN N.V. (Dutch fixed-line and mobile telecommunications company)
LAN	Local Area Network
LCD	Liquid Crystal Display
MAC	Medium Access Control
M-MGw	Mobile Media Gateway
MMOG	Massively Multiplayer Online Games
MMORPG	Massively Multiplayer Online Role Playing Games
MPE	Multi Protocol Encapsulation
MPEG	Moving Picture Experts Group
MSC-S	Mobile Switching Centre Server
MTU	Maximum Transfer Unit
MVDS	Multipoint Video Distribution System
NACK	Non-Acknowledge (Packet)
NFC	Near Field Communication
NFZ	Polish National Social Security Services
NHK	Nippon Hōsō Kyōkai (Japan Broadcasting Corporation)
NTT	Nippon Telegraph and Telephone (Corporation)
NTT-DoCoMo	NTT - do communications over the mobile network
OBS	Optical Burst Switching
OCS	Optical Circuit Switching
OECD	Organisation for Economic Co-operation and Development
OLT	Optical Line Terminal
OMNIMAX	Variation of the IMAX format
OPEX	Operational Expenditures
P2PTV	Peer to Peer Television
PDA	Personal Digital Assistant
PC	Personal Computer
PER	Packet Error Rate
PET	Petrol Equivalent Ton
PIN	Personal Identification Number
PLC	Power Line Communication
PLMN	Public land mobile network
PLR	Packet Loss Rate
PON	Passive Optical Network
POTS	Plain Old Telephone System
PSNR	Peak Signal-to-Noise Ratio
PSTN	Public switched telephone network
PSTS	Public switched telephone system
QoE	Quality of Experience
QoS	Quality of Service
RBS	Radio Base Station
RDP	Reliable (User) Datagram Protocol
RFID	Radio-frequency identification
RPG	Role Playing Games

RTP	Real-time Transport Protocol
RTCP	Real-Time Control Protocol
RTS	Real Time Strategy
RTSP	Real Time Streaming Protocol
RTT	Round Trip Time
RX	Receiver
SBR	Spectral band replication
SMATV	Satellite Dish (TV)
SMS	Short Message Service
SOHO	Small Office and Home Office
TCO	Total Cost of Ownership
TCP	Transmission Control Protocol
ToIP	Telephony over IP
TS	Transport Stream
TX	Transmitter
UDP	User Datagram Protocol
UHTV	Ultra High Definition Television
USB	Universal Serial Bus
UWB	Ultra Wide Band
VBR	Variable Bit Rate
VDSL	Very High Speed Digital Subscriber Line
VLAN	Virtual Local Area Network
VNC	Virtual Network Computing
VoD	Video on Demand
VoIP	Voice over IP
VR	Virtual Reality
WiFi	No signification – wireless interface
WLAN	Wireless Local Area Network
WP	Work Package

2 Methodology

2.1 Classification of services

The communication networks essentially allow an exchange of information between persons, between persons and equipment (e.g., a video server), and between equipments (e.g., a sensor and an actuator). Based on the type of the information exchange and the inherent service requirements, we can identify the following groups/classes of services which will serve as a basis for the services identification in the rest of this document:

- Basic communication such as telephony, e-mail, and instant messaging
- Internet-related services such as general browsing, e-banking, e-shopping and similar; including files sharing
- Video-related services such as Video on Demand , IPTV, video conferencing[1] and similar
- Online Virtual Environments such as social network or gaming
- Remote Technical services such as the ability to remotely control/survey your home
- Remote Health services such as remote health monitoring.

2.2 Common Metrics Definition

The service requirements will be assessed in terms of the following common metrics:

- Bit-rate/Throughput (Minimum Reserved/Maximum Sustained Rate), three classes are considered: Constant Bit Rate (CBR), Variable Bit Rate (VBR) and Available Bit Rate (ABR).
- Delay (Maximum Latency Tolerance and Delay Variations). This integrates the Packetisation (encapsulation) delay, the Network transit delay and the Reconstruction delay.
- Jitter (Jitter Tolerance = Delay variations acceptable). Jitter can be compensated for by a buffer at the price of an additional delay.
- (Tolerable) Packet loss
- Mobility: impact of mobility requirements on the service requirements. This is naturally limited to a user moving from one place to another within a building or its direct vicinity (limited mobility, no cars etc…).
- Traffic Priority (Guaranteed service, Controlled load or Best effort)
- Security

A service may have different requirements depending on the protocols which will transport it. These differences will be highlighted where applicable. At the end of each section a summary table is provided.

[1] Videoconferencing can also be identified as a basic communication service. Technologically, it is however close to the video-based services and thus put into this group.

3 Basic Telecommunication Services

3.1 IP telephony (Voice-over-IP)

Internet Protocol (IP) telephony (sometimes referred to as Voice-over-IP or simply VoIP) is the ability to phone over the broadband access network as opposed to using the conventional (twisted copper pair) telephone network or POTS (Plain Old Telephone System). IP telephony has attracted many residential users mainly because the price of the communication is much lower than conventional fixed telephone lines and does not depend on the distance of the call. This applies as well for enterprises, attracted by the stronger ability to manage calls when they are IP-based. In the issue of "Telecom Strategies and Trends" (a journal published by InfoCom) from May 2008, the decline of POTS compared to VoIP is estimated in Europe: in 2006, 6% of all calls were VoIP ones, this proportion rose to 11% in 2007 and is forecasted to be around 22% by 2012 [1]. In the US, there are currently (1Q08) 17.4 million VoIP subscribers representing around 14.8% of the total number of households. This number is 90% higher than for 1Q07 [2]. The VoIP service is now an essential part of the pool of services customer require from their Internet Service Provider.

IP telephony as well as many other multimedia applications with real time constraints uses the User Datagram Protocol (UDP) as the transport layer protocol. The transmission bandwidth necessary for VoIP can be found depending on the different codecs that are used such as G.711 or G.722.x, they vary from 12 to 96 kbps. The overall one-way delay is recommended to be lower than 400 ms (ITU-T G.114) and a delay shorter than 150 ms is considered to be imperceptible. In terms of jitter, the specified or recommended limit values are harder to define as packets are re-synchronised by the decoding equipment jitter buffer. After resynchronisation, the jitter must be below 1 ms (ITU-T G.1010*). In general, jitter values lower than 50 ms are recommended. In terms of Packet Loss, a 3% value is considered as the highest limit for voice. However, a value of 0.1% must be targeted as the maximum value to allow for video transfers.* These values may also depend on the Packet Loss Concealment algorithms used in the chosen codec.

3.2 E-mail

E-mail, together with web-browsing, is one of the “oldest” and most used basic services among the Internet users. The main requirement is error free transmission. The delay to receive an email is not critical and can be in a range of several seconds [3]. Taking into account the large space that email can reach (up to a few Megabytes with attachment, up to hundreds of kilobytes without), it is necessary that the network transport rate is in the region of (at least) 1 Mbps for downloading email into a computer (i.e. with their attachment) or in the region of 10 to 50 kbps for a mobile terminal when only the email text is downloaded (without the attachments).

3.3 Short Message Service (SMS) and Instant Messaging

The idea that the SMS (Short Message Service) would be used for sending text messages between mobile users is one which few believed to become true. However, finally SMS has become one of the most extended and profitable services in the mobile telecommunications industry. For instance, it is estimated that 71.6 % of all French mobile phone users send at least one SMS per month and 15 % send more than 50 SMS per month [4]. SMS is still growing strongly in Europe [5]. The SMS capability has even been extended to conventional telephones.

The requirements are not high; in fact, it is enough if the message arrives within a few (or even a few tens of) seconds. SMS was created as part of GSM, but nowadays, the messaging exists on a wide range of networks, including 3G networks, and it is able to include not only text but also multimedia files attached.

Instant messaging is a similar service, but taking place on the real-time and over a fixed infrastructure (most often, over the Internet). The existing implementations are numerous (Jabber, Windows Life Messenger, ICQ, Skype and many others). The service usually incorporates a search function allowing to look for chat partners. The requirements are very relaxed (by fixed network standards) since the amount of data transferred is very small: the delay must be in a sub-second region.

3.4 Summary table of basic telecommunication services

Table 3-1 summarises the requirements for the basic communication services.

Table 3-1: Summary for basic communication services

Service	Bit rate	Delay	Jitter	Packet Loss	Mobility	Traffic Priority	Security
IP telephony	12 – 96 kbps	< 400 ms, 200 ms recommended	< 50 ms	< 3% < 0.1% recommended	Yes	High	No
E-mail	1 Mbps/ 50 kbps (mobile)	Relaxed specification (seconds)	Relaxed specification	None (BER < 10^{-6})	Yes	Low	No
SMS and Messaging	Low (10 kbps)	Relaxed	Relaxed specification	None (BER < 10^{-6})	Yes	Low	No

4 Internet Based Services

4.1 Internet browsing

In contrast to VoIP, there is usually no real-time constraint associated with data transfers related to Internet browsing. The protocol used is often the Transmission Control Protocol (TCP; in almost all cases over IP) which offer different characteristics than UDP.

For residential users, Internet browsing is one of the most common services. Also for mobile phone users, it is a much-used service [6]. With the browsing programs, the users explore/navigate all Internet information. This information can be of diverse formats (text, video, audio, pictures, motion video). This is an interactive service that needs a delay inferior to 4 seconds per page, no frame error rate is allowed, the bit error rate must be up to 10^{-8} and the expected jitter must be less than 10 ms [3].

4.2 Music streaming

The main requirement with respect to this streaming service is the quality of the sound perception for the end user. So, it is possible to go from a low quality service (mainly only vocal) with a data rate of 5 kbps to a CD sound quality that needs 128 kbps. The initial delay will be less than 10 s (to start listening to the stream), and the PER<1% [3].

There are many different players on the market to provide this service. Most of them use RTSP (Real Time Streaming Protocol) which is based on RTP (Real Time Protocol).

4.3 File sharing using peer-to-peer

Peer-to- peer file sharing traffic has been evaluated today has the main contributor to bandwidth usage in the world. For instance, in North America, in 2008, it is estimated that 35% of all downstream traffic (from the network to the user) will arise from the peer–to-peer file sharing while, in the upstream direction, this proportion rises to 75 %. The overall percentage (upstream plus downstream) shows a total of 44% of all internet traffic being linked to peer-to–peer file sharing, an increase of 4% compared to 2007 [7].

A peer-to-peer network architecture is in general a technical solution, which can be used for a number of applications. File sharing is probably the most famous application of peer-to-peer networks. Examples of the file sharing networks include KaZaa, Gnutella, and BitTorrent [8]. Despite the popularity of the file sharing, potential applications of peer-to-peer networks stretch far beyond that. Namely, the peer-to-peer approach can be (and is) used for distributed computed, distributed storage, distributed search engines, Internet telephony (e.g. Skype), TV distribution (P2PTV, e.g. Joost), steaming media, and others.

Since the new users joining the network bring their own bandwidth which is then shared between all the users, the peer-to-peer networks are perfectly (and easily) scalable. An example of such a network is Skype with its (currently) 200 million users [8]. The bandwidth sharing also means that a user can receive data from multiple users up to his/her download capacity (the same applies to the upload). As the matter of fact, peer-to-peer applications are perhaps the ones of the very few applications *today* which can in principle exploit all the available end-user access bandwidth.

The network requirements for peer to peer are similar to those of internet browsing (see section 4.1).

4.4 Web3D

The next major application development in the Internet's evolution can be Web3D, which is a system of linked interactive 3-D and 2-D environments that will include everything from user-specific, private applications like immersive learning simulations to virtual worlds open to anyone who wants to join. People will move among these in a seamless, natural way. Web3D will deliver an interactive, immersive experience that increases motivation and engagement compared with the static, text-oriented or even somewhat interactive graphical interfaces of today's Web. People will be represented visually by avatars that can move in space and communicate with others via voice and text, gestures, user-directed motion, animation sequences, and social networking tools. Web3D will integrate with Web 1.0 and Web 2.0 tools and technologies, as well as business software applications [9].

4.5 Summary table of Internet-based services

Table 4-1 summarises the requirements for the Internet-based services.

Table 4-1: Summary for Internet-based services

Service	Bit rate	Delay	Jitter	Packet Loss	Mobility	Traffic Priority	Security
Internet	1 - 100 Mbps	Relaxed specification	< 10 ms	None (BER<10^{-8})	Yes	Low	No
Music	5 – 128 kbps	Buffer dependent	Buffer dependent	< 1%	Yes	High	No
File sharing (peer-to-peer)	1 Mbps - 100 Mbps	Relaxed specification	< 10 ms	None (BER<10^{-8})	Yes	Low	No
Web3D (future)	10 Mbps to 1 Gbps	Relaxed specification	< 10 ms	None (BER<10^{-8})	Yes	Low	No

5 Video-Based Services

5.1 IPTV

Internet Protocol-based television, or simply IPTV, together with Video-on-Demand (VoD) is the third service part of the conventional triple play service usually offered by telecom operators/Internet Service Providers (ISP). Its goal is to bring live TV to the end-user (otherwise obtained from a TV aerial, satellite dish or cable) via an IP connection. This service is a high revenue generator and is often used as a differentiator between the different ISP to attract new customers. Worldwide, there are well over 7 million IPTV subscribers [10] and the situation for IPTV is one of great success [11]. For example, Swisscom has around 70,000 customers for its BlueWin TV service (growing by 2,000 new customers per week), Portugal Telecom has 100,000 IPTV customers (the service is called MEO), and this number has grown by 53,000 in the last quarter, Belgian telecom operator Belgacom had 349,000 customers for their IPTV service at the end of the first quarter of 2008, representing an increase of 44,000 customers during that quarter, France Telecom / Orange has 1,149,000 IPTV customers [10], TeliaSonera, British Telecom, Deutsche Telecom, Telekom Austria, KPN, RomTelecom (Romania) and more are all in the process of offering or improving their own IPTV service. As well, China Telecom is currently selecting a manufacturer to build 574,000 IPTV Set-Top-Boxes. (more examples can be found in [11]).

To provide IPTV, the different channels are first compressed using MPEG-2 or MPEG-4 (H.264/AVC [12] codecs) and are subsequently streamed to the end-users via multicast using the Internet Group Management Protocol (IGMP) and based on RTP, a generic Transport Protocol for Real-Time Applications. RTP also provides tools for synchronization (RTP time stamp, RTCP Real-Time Control Protocol).

For mobile telephony users, the TV service is specifically developed for 3G mobile phones. It is a real time streaming service that also uses RTSP over UDP to minimize the influence of the delays. The packet loss allowed is higher than if using TCP but this phenomenon has less influence on this kind of services. The expected data rate with this service is 128 kbps and with a final resolution of 176x144 pixels and 15 Hz. The next step with this streaming service for 3G mobile phones is to upgrade the data rate to 384 Kbps and the resolution to 176x144 pixels and 30 Hz or 352x288 and 15 Hz. More into the future, a higher quality version of this streaming service is to be expected with a data rate from 2 to 10 Mbps. This service would target 3G mobile phones, handheld devices and TV sets.

Orange Innovation announced [13] that 3D-TV will be offered in the near future (end 2008) to 3G+ enabled mobile devices. The screen of the device has to be compatible with 3D viewing (special glasses will not needed) and the data rate required for the video stream is between 50 and 90 Mbps.

5.2 Video-on-Demand (VoD) and Multimedia content production and delivery

Video on Demand uses the Real Time Streaming Protocol (RTSP) to stream the video file to the client after compression of the signal using again MPEG-2 or MPEG-4. In order to do this, similar requirements to IPTV are expected. However, the streaming of the video file is only a part of the VoD service, and the actual recording/transfer of the file on a local server to allow the end user to play it locally afterwards is also to be considered. Because of the typically large size of the video file (up to a few GB), it is required that the network can offer sufficient bandwidth so as to allow a file transfer in as short a time as possible (less than one minute). An available bandwidth of 1 Gbps would enable a video download in a few tens of seconds and consequently make the service offer suitably quick and pleasant to use.

5.3 Video conferencing and video telephony

There is an expectedly big market in the future for a service offering video conferencing of high quality to the end-users. Already people use video telephony/conferencing services using a USB camera in conjunction with various software (e.g., Skype [14] and some others) even though the image quality and the synchronisation between the sound and image depend on the channel quality (bandwidth, delay, jitter) and can be quite poor for narrowband connections. The quality factors in a good video-conferencing system include the following (by the order of decreasing importance): the delay in the sound and picture transmission must be unnoticeable (comparable to that of VoIP requirements), the synchronisation of the sound and image at each end must be near perfect (i.e. < 1 ms) and the image of the remote speaker must be of an acceptable size (comparable to the resolution of a standard TV set).

To comply with these demands, it is necessary to encode the images (compress) so as to minimise the data flow volume. The codecs usually employed for this are H.261, H.263, H.263+, H.26L, and so on. Another family of codecs more dedicated to video conferencing are the H320, H323 and H324, although H323 is the most extended one.

Note that a medium or low image quality can be accepted by the user, the main requirement is on the delay (the time it takes for the image to travel between the users) and the synchronisation between image and sound. To offer real-time interactivity the end-to-end delay should be kept less than 150 ms, a quasi-real-time condition is satisfied if the end-to-end delay is kept below 400 ms (acceptable for long distance communications). If the delay exceeds this threshold, the users tend to accommodate to a half-duplex communication. The frame data error (FER) must be less than 1%.

However, as the bandwidth available will increase in the access network, the quality requirements will most likely become stronger. It is probable that users will also accept the fact that both the comfort and quality will be different depending on the environment and the equipment used (mobile, TV set, dedicated video conference room). This tendency has already appeared. Indeed, today the necessary flows are from 128 kbps (2 ISDN lines) to 2 Mbps [15] for a projection on computer screens or television sets. But, new video conference services offered by the operators such as "*Realmeet*" which has been marketed by France Telecom [16] since December 2005 allow the projection of a real life size distant interlocutor, ensuring both eye contact and spatialization of the sound in connection with the position of the interlocutor. This comfort is probably necessary for a significant development of the video conference applications. It should be noted that in this case, the necessary bit rates are then of the order of 4 Mbps.

5.4 Video Streaming / Home Theatre

In this section, we refer to the distribution within the home of an uncompressed video stream from a media server or player (DVD for instance) to one or several media renderers (TV set for instance). The fact that the video stream is uncompressed means that no special coder/decoder is needed at both ends of the transmission and this situation is identical to the replacement of a cable between two video/multimedia end devices which are situated in the same or in different rooms.

For this application, we will consider the High Definition Media Interface (HDMI) standard as many (most, if not all, in the future) video devices are equipped with it. Indeed, there is today more than 700 manufacturers of HDMI equipped end-devices and it is estimated that, in 2008, more than 200 million HDMI enabled devices will be shipped, leading to a forecast of more than 1 billion HDMI devices in use by 2010 [17]. HDMI can transmit video and audio signals of all formats (standard to high definition). Depending on the video format to be transmitted (number of pixels, colour coding depth and image refresh rates), the data-rate requirements vary. All HDMI interfaces adhere to a standard from HDMI 1.0 to HDMI 1.2 and more recently to HDMI 1.3. All standards can transport a video stream of different formats plus a number of audio streams. On top of this, some control information is exchanged between the RX and TX.

The HDMI Specification [18], shows that the data rate needed to transport the HDMI flow is primarily dictated by the video format and colour coding used. As well, additional coding is used to increase the performance of the transmission (blanking, BCH forward error correction plus 8B/10B encoding). The final aggregate data rates go from several Mbps (VGA format or 640 x 480p @ 59.94/60Hz) to 10.2 Gbps (in the case of the so called "FullHD" TV or 1920 x 1080p @ 59.94/60Hz with 48 bit "Deep Colour"). The maximum error rate tolerated is 10^{-9}. *In the future, formats like Ultra HDTV also* *known as "4K" having a 7680 x 4320 screen resolution are envisaged, leading to compound data rates around 24 Gbps* [19] (see as well below).

5.5 Immersive TV

Conventional TV systems, even the HDTV standard, cannot provide an immersive experience, due to the size limits of the nowadays available TV screens. A way to overcome the size limit is the use of projectors; however, in this case the number of pixels per inch is largely reduced, thus limiting the resolution of images and the feeling of reality of the viewer.

An example of immersive view is the IMAX Dome/OMNIMAX cinema system developed in 1960 (see, for a brief introduction, [20]), where the images are projected on a dome that allowed the viewer to watch an image extending 180° from left to right, 110° above and 20° under the horizon; the equivalent resolution of such a view is roughly 10k times 7k pixels. It is possible then to propose a home solution comparable to the IMAX Dome, projecting the image on a “dome” that is roughly a quarter of the surface of a sphere. The surface that must be covered by the image is simply πR^2 where R is the radius of the dome itself; taking as a reference a 61" screen, whose surface is approximately 1 m^2, it is possible to say that the previous formula represents the number of 61" screens needed to “cover” the dome. Assuming for each screen a 1080p bit rate (H264, High Profile, level 4.2 [12]), the equivalent bit rate of a dome screen is calculated to be 812.5 Mbps for a 2 m radius screen (equivalent to 13 61" screens) and 7125 Mbps for a 6 m radius screen (equivalent to 114 61" screens). Note that this bandwidth requirements are the maximum requirements defined by the standard.

An approach similar to the dome one above mentioned, that makes use of a more classical flat screen instead of a dome for the projection, was proposed by Nippon Hōsō Kyōkai (NHK – Japan Broadcasting Corporation) introducing the Ultra HDTV (UHTV) in [19]. The screen resolution of UHTV is 7680x4320 pixels (the format is progressive, not interlaced), 16 times bigger than a standard HDTV screen. The bit rate of the uncompressed images is 24Gbps: an experimental verification of optical transmission of an uncompressed signal was performed by NHK in 2006 [21], using already deployed dark fibres and a DWDM wavelength allocation. More recently, in 2007, NHK developed a codec for both video and audio compression in order to squeeze the 24 Gbps into 600 Mbps [22]. According to NHK previsions, this solution will be suitable for broadcasting in 2025.

Similar approaches have been proposed in recent years by HHI [23] but the requirements in terms of bandwidth are fairly similar for the same quality and function.

The proper delay itself turns out not to be a very strict requirement for this kind of applications (without real-time interactivity) [24]. Jitter (delay variations with time), however, can be the major cause of loss of video quality at the receiver for packet-based networks [24]. To overcome these impairments, the usual way is to introduce a buffer at the receiver to reconstruct the correct sequence of the received packets and assure the appropriate playback timing [25]. It is intuitive that the buffer size introduces a further delay into the system. As for the packet loss requirements, an analysis of the effects of packet loss on video streams can be found in [26]; a possible solution for the packet loss recovery for MPEG-4 (or for a video coding technique that uses I, P and B frames) can be found in [27], where the effects of error propagation due to packet loss are exploited.

5.6 Immersive video conferencing

A description of a system capable of providing immersive video conferencing services is provided in section 5.5, which is devoted to the immersive TV; al long as the main differences between the two systems are related to the application, but not to the architecture, please refer to that section for the description. It is reasonable to assume that the amount of information that needs to be transmitted is lower than in an entertainment scenario; thus lower bandwidth requirements can be considered. The main difference, however, will be the delay. As stated before, to provide real-time interactivity the end-to-end delay should be kept below 150 ms; this value includes also the jitter compensation buffer. However, if the delay exceeds this threshold, users tend to auto compensate by commutating spontaneously into an half-duplex communication [28][25].

5.7 TV-Broadcast (DVB-x)

DVB, short for Digital Video Broadcasting, is a suite of internationally accepted open standards for digital television. DVB systems distribute data using a variety of approaches, including by satellite (DVB-S, DVB-S2 and DVB-SH; also DVB-SMATV for distribution via SMATV); cable (DVB-C); terrestrial television (DVB-T, DVB-T2) and digital terrestrial television for handhelds (DVB-H); and via microwave using DTT (DVB-MT), the MMDS (DVB-MC), and/or MVDS standards (DVB-MS). These distribution systems differ mainly in the modulation schemes used and error correcting codes used, due to the different technical constraints. Important in the context of this work is in particular the DVB-IP (Digital Video Broadcasting over IP) standards such as ETSI TS 101 154 [29], ETSI TS 102 154 [30] and ETSI TS 104 468 [31], which specifies the transport of video and audio services over IP-networks.

It is expected that in certain areas DVB services will be delivered and used in parallel and simultaneously with video broadband services. For instance, Swisscom offers an IPTV service via its broadband access network (called BlueWin TV). Since May 2008, this service has now been extended to mobile phones (BlueWin TV Mobile) using DVB-H technology [32]. Thus it is likely that DVB services have to be integrated in the home network architecture and they have to be considered in the entire bandwidth and performance consideration. DVB services will be used as real time video-audio data streams and they will also be stored on PCs and home servers. In advanced scenarios DVB services might also be used in bidirectional way, e.g., in typical rural areas with under developed broadband coverage.

For DVB-IP, a protocol stack is defined in a suite of DVB specifications. The transport of video and audio data is based on RTP, which also provides tools for synchronization (RTP time stamp, RTCP Real-Time Control Protocol). Video is encoded/decoded according to the H.264/AVC or VC-1 standards and audio is encoded/decoded according to the HE AAC v2, AMR-WB+, AC3 or E-AC3 standards [33]. The MPEG-4 and MPEG-2 standards provide the necessary information to set out the requirements for DVB-IP transport. The IP network shall comply with the mandatory network requirements to guarantee successful delivery and decoding, as defined in [34] for MPEG-2 Transport Stream. The most important ‘mandatory’ requirement has to do with the packet jitter, which is defined as the variation in delay between the source of the stream and the end device. This jitter should be maximum 20 ms or thus 40 ms peak-to-peak. Two ‘recommended’ constraints for these IP networks relate to the packet loss and the multicast timing. Packet loss should incorporate maximum one noticeable artefact per hour. For a 4 Mbps transport stream with seven transport stream packets per IP packet, one error per hour is equivalent to an IP packet error rate of less than 10^{-6}. When AL-FEC is used then the acceptable IP packet loss rate may be considerably higher [34].

5.8 Innovative TV services: stereoscopic (3D) and free viewpoint TV

This section is intended to provide a description of some innovative TV services, such as stereoscopic (3D) and free viewpoint TV, that will be likely be available in the future.

Recently, the interest for 3-Dimensional (3D) applications has strongly increased, leading to different projects sponsored by the European Community: among the others, ATTEST (a project of the 5th framework program [35]), coordinated by Philips, aimed to develop a complete broadcast chain for 3D content, and 3DTV (a project of the 6th framework program [36][37]) led by Bilkent Universitesi (Turkey) which is a Network of Excellence aiming to cover all the topics concerning 3D TV in an integrated manner.

In Framework Programme 7 (to which the ALPHA project belongs), a number of projects will also deal with topics related to innovative TV services (including 3D). Those include the 3D4YOU project [38], coordinated by Philips, that will focus on 3D TV; beside this one, the HELIUM3D project [39] led by De Montfort University, aims to develop a technology suited for an auto stereoscopic display with no limitation to the number of users (to achieve this result, a pupil position tracker is planned to be used). The project REAL 3D [40] is focused on a real 3D transmission based on dynamic holography; this, as reported in [41], is apparently the ultimate frontier of 3DTV today.

The innovative TV services (described in more detail below) can be summarised as follows:

- Stereoscopic TV includes different ways to recreate a 3D representation of the images, in order to transmit at the viewer the correspondent feeling of depth, with images not only on the screen plane, but also ahead and behind the screen.
- Free viewpoint TV: in this TV service, the point of view of the show (a sport event, a play, a concert) can be freely chosen by the final user (the viewer). Similar services are already provided from different broadcasters, but limited to views correspondent to different physical cameras used for the broadcasting; the aim of such a service is to provide to the user the capability to freely choose any possible viewpoint, not necessarily corresponding to a physical camera.

Stereoscopic (3D) TV:

Stereoscopic TV is a way to provide the final user a system that is capable to recreate the perception of the depth of the images (perception of depth is a consequence of binocular vision in human beings), thus achieving the goal of allowing 3D vision of 2D images.

A possible way to do so is via the Pulfrich effect [42] in which an optical illusion of depth is achieved via well defined camera movements and providing the user a special pair of glasses (with one lens darker than the other). The only pro of such a technique is the bandwidth required is the same of a normal TV broadcast. The cons are wide and many (mainly, low image quality and restricted application range, in particular, no still images).

A large amount of research work ([43][44][45][46]) was developed by RAI (Radio Televisione Italiana, the public Italian television company); in the past, several projects funded by the European Community, focused on stereoscopic television, such as COST 230 [47], DISTIMA [48], MIRAGE [49].

Recently, commercial solutions capable of achieving stereoscopic vision made their appearance on the market: a first solution, based on the Digital Micro-mirror Device (DMD) by Texas Instruments, consists in an LCD monitor/TV developed by Samsung [45][50] that is capable of alternatively showing the two different images (left eye / right eye) that are needed for stereoscopic vision via a tilting mirror. With such a technique, a device that alternatively covers the user's eye, properly synchronized with the display, is needed (passive or active shutter glasses).

A second solution, proposed by Philips [51], is based on an *auto stereoscopic* monitor [44]; this device is covered with a set of cylindrical lens that are designed to irradiated the light emerging from the screen in different directions, thus the images that hit the eyes of the viewer are different. With this technology, no head mounted devices are needed for the correct stereoscopic vision (since the name auto stereoscopic).

Considerations about bandwidth occupation required by such techniques: in each case, the screen is assumed to be 1080p (1920x1080 pixels, progressive; see, for instance, [52]) and the video is coded via H264 [53][54]. Bandwidth requirements for stereoscopic view can be similar to that of conventional video leading to 62.5 Mbps (high profile, level 4.2 [12]). For some schemes (e.g. pure stereoscopic and auto-stereoscopic), both the images (the left-eye and the right-eye ones) need to be transmitted and processed by the viewing device. In order to keep the format progressive, a doubling in the frame frequency is then needed, thus 300 Mbps (high profile, level 5.1 [12]).

A different approach, developed by Philips and called '2D-plus-Z' (see [55]) consists in the transmission of a single 2D image plus a grey-scale image of the same dimension containing the mapping of the depth of the 3D desired image; also in this case, the viewing device should be capable of performing the required processing. This technique is convenient in terms of bandwidth occupation, due to the fact that the transmission of the depth map only increase the bit rate of the original stream by about 5-20%.

Free view-point TV:

Another innovative service that can be offered in the field of Home TV, is the possibility of select the users viewpoint of a show (this applies in a most natural way to live events). This approach differs from the immersive TV: in that case, in fact, the viewer's position inside the scene is fixed, with just the possibility of watching at a different part of the screen. Such a service is, to provide few examples, the aim of the DTI-iview project promoted by BBC (British Broadcasting Corporation) [56][57][58][59] and of the IVVV project by HHI [60].

For this kind of services, even if it is largely different from the immersive TV via head-mounted devices discussed in 5.5, in terms of bandwidth occupation similar considerations can be made: in this case, in fact, the user is allowed to select the desired view point that in principle is similar to the previous case in which the user was able to "look around" at the picture, and the same scenarios emerge: in the first one, again, the viewpoint is communicated by the user to the broadcaster via a dedicated uplink, and the broadcaster in turn sends the appropriate view to the user; the computational effort is done by the broadcaster, the bandwidth requirement is relatively modest (62.5 Mbps in downstream, plus the small uplink channel).

In the second one, the broadcaster transmits all the required information and the elaboration of the required view-point is made locally at the user's side. In this case, the bandwidth allocation could be many times larger than the previous case. To provide an example, in [56] a set of 15 cameras is devoted to the representation of a single portion of the scene (in this case, a football field), thus the maximum required bandwidth assuming that each camera broadcasts an 1080p stream, is 15*62.5=937.5 Mbps.

An approach suggested by HHI in [60] for lowering the bandwidth requirements for live events (i.e., for situations with a well defined and stable background, such as a stage, or a stadium) is to pre-record and pre-broadcast the data concerning the structure, in order to allow the user to locally download a part of the data required for the elaboration not in real time, thus relaxing the bandwidth allocation (the real-time data will concern only the "non-static" part of the event, such as players or actors).

5.9 Summary table of video-based services

Table 5-1 summarises the requirements for the video-based services.

Table 5-1: Summary for video-based services

Service	Bit rate	Delay	Jitter	Packet Loss	Mobi lity	Traffic Priority	Secu rity
IPTV	2 – 20 Mbps (for HD)	< 400 ms; 200 ms recommended	< 50 ms	< 1% < 0.1% recommended	Yes	high	no
VoD	Min. 2 Mbps Future: 1 Gbps	< 400 ms; 200 ms recommended	< 50 ms	< 1% < 0.1% recommended	Yes	high	no
Videoconference	128 kbps to 4 Mbps	< 400 ms 200 ms recommended	< 50 ms	< 1% < 0.1% recommended	Yes	high	no
Video Streaming (uncompressed)	Min. 128 Mbps. Recommended: 1 Gbps Future: 10 Gbps	< 400 ms 200 ms recommended	< 50 ms	< 1% < 0.1% recommended	No	high	no
TV broadcast (DVB-IP)	96 kbps to 45 Mbps (HD)	< 400 ms	< 20 ms	None (or use FEC)	Yes	high	no
TV broadcast (DVB-x / non IP based)	N/A rather BW occupied up to 8 MHz	< 400 ms	< 20 ms	None (or use FEC)	Yes	high	no
Immersive TV (future; using UHDTV)	24 Gbps uncompressed. < 640 Mbps compressed	< 400 ms; < 150 ms recommended	< 20 ms	< 0.4%	No	high	no
Immersive Videoconference (future; using UHDTV)	< 640 Mbps compressed	< 400 ms; < 150 ms recommended	< 20 ms	< 0.2%	No	high	no
Stereoscopic TV (future)	Min. 62.5 Mbps, up to 320 Mbps	< 400 ms; < 150 ms recommended	< 20 ms	< 0.4%	No	high	no
Free Viewpoint TV (future)	937.5Mbps	< 400 ms; < 150 ms recommended	< 20 ms	< 0.4%	No	high	no

6 Online Virtual Environments

6.1 Massively Multiplayer Online Games

Massively Multiplayer Online Games (MMOGs) are a prime example of Online Virtual Environments and have become one of the potential “killer” applications of the Internet. Since their appearance in the eighties, the customer base for Massively Multiplayer Online Games has grown from a few thousands to tens of millions of clients. For one instance of such an application, the level of concurrency has also increased from a few dozens to tens of thousands of clients interacting simultaneously with each other and the virtual world. Examples of these applications include World of Warcraft [61] and Second Life [62], the former currently having over 10,000,000 paying users with peak rates of over 500,000 players online at the same time albeit distributed over multiple parallel worlds called realms.

MMOGs and in particularly Massively Multiplayer Online Role Playing Games (MMORPG) [63] are a genre of computer games in which a large number of players interact with one another in a virtual world. The movement of objects in MMOGs is usually recorded as a vector, meaning that motion is recorded as a combination of magnitude and direction—in other words, the server transmits information to all the clients in an area that various objects are moving in a certain direction at a certain velocity. This works better than a system that only transmits absolute coordinates which would require much more bandwidth to generate a fluent movement of players in the virtual world.

Because MMORPGs are large virtual worlds, and because it would consume massive amounts of bandwidth to report to every client what happens with every object and event within the virtual world, techniques have been created to divide the virtual world into discrete units (often called “zones”, “cells” or “shards”) and allocating different servers to different areas of the game as well as separate servers for dungeon "instances." Chat servers provide communication between players. Although players might be connected to different game servers depending on where they are in the virtual world, things such as a guild-chat or instant messaging needs to work across the entire game. In 2006, Vivendi reported that World of Warcraft, the most famous MMORPG, maintains over 1900 servers globally [64].

When we look at the domain of online games, the required bandwidth is hardly ever the bottleneck. Current games are often developed to have the clients not require more bandwidth than the narrowest last mile link (until recently a 56K modem) [65].

Regarding the delay of online games, its impact can differ between game types. More specifically, in [66] a distinction is made between games with an avatar, and omnipresent games. In the first categories, the authors advise a threshold of 100 milliseconds for first-person games such as racing and first person shooters (FPS); and a value of 500 milliseconds for third-person games such as sports games and role playing games (RPGs). In the latter category, the authors argue the bound is up to 1000 milliseconds. An evaluation of performance versus latency for different classes of online games is found in [66].

In Technical Report TR-126 of the DSL Forum [67], a target round-trip time (RTT) of 100 ms is mentioned for a generic game type. This corresponds to the value recommended for typical first person shooters (FPS) like Quake3, Unreal Tournament, Half Life 2 and racing games. Other studies [68][69][70] reach similar conclusions, but focus on individual games. The authors of [71] have observed that packet loss values as high as 5% are hardly noticed by the players. In [72] the impact of packet loss and delay on two games, Quake 3 and Halo is studied. The authors conclude that games should only be played over the Internet when the designers incorporate measures to react to delay and loss. They also noticed that while for Quake 3 no significant impact on the Quality-of-Experience (QoE) was seen for packet of up to 35%, the Halo players already noticed a great decrease for even a small amount of packet loss. The decrease in QoE was reflected by a greater number of players wanting to leave the server. In [73] the experiment was repeated for Halo2, the successor of Halo. The authors noticed that the new incarnation of the game was much more resilient to packet loss.

6.2 Online Distributed Virtual Environments

Online virtual worlds will no longer be limited to games, Second Life already shows that it can be a place of business or social networking community. Online virtual worlds will also get input from projects such as Google Earth [74] and Microsoft Virtual Earth 3D [75][76] which present a virtual 3D model of the earth and allow users to add and create additional content like 3D images of buildings. It is not unlikely that users of services like this will be given the ability to travel in the virtual earth using avatars and are able to communicate with each other, very similar to the experiences offered by current MMOGs but with a focus on social networking or online business instead of online gaming. Looking at the growing success of social networking sites such as MySpace [77], Facebook [78] and Netlog [79] it is to be expected that the number of users participating in online virtual and social worlds will only increase.

Current business offerings related to MMOGs and Online Distributed Virtual environments focus on the improvements of software and middleware architectures (enabling technologies), like Big World [80]. However, future developments will more likely focus on the integration of available technologies and virtual environments and communities. For social networking sites, initiatives like OpenSocial [81] are proposed and supported by companies like Google, MySpace and Yahoo!. OpenSocial defines a common API for social applications across multiple websites. Similarly initiatives for supporting Open Source and/or portable avatars across virtual environments are proposed by IBM and Linden Labs [82][83].

The integration of these existing (web-) technologies and online communities and environments with each other and new applications will be one of the main challenges in the near future. Currently live events are already streamed in virtual worlds like Second Life (e.g. concerts or tv-shows are broadcasted as well in the real world as in the virtual world). Virtual events are broadcasted into the real world through online radio stations or tv-broadcasts etc.

Requirements imposed on the network level by these new applications of distributed virtual environments are difficult to assess.

- User to application: probably similar to current applications when it comes to latency and jitter. Bandwidth will probably be a lot higher because of higher quality content, and a significant amount of user-generated content.
- User generated content means that the actual content cannot be installed locally and rendered at the local PC but will either have to be downloaded or streamed (live). Streaming is the model currently already employed by Second Life:

  > "Each computer running Second Life will need an average of 80 kbps downstream, spiking at about 400 kbps on initial connect and during teleports. Upstream is much lower, requiring 30 kbps on average. Make sure your network connection can handle these bandwidth requirements if you plan to run Second Life on multiple computers at the same time." [84]

Communication between virtual worlds, online communities etc. will have certain requirements as well of course.

An example of future latency requirements are virtual reality (VR) environments. In this case the latency means the time that elapses between the event where a player requests an experience and the event where a player sees and hears that experience on his or her computer screen [85]. Meehan et. al. reports effective VR environments shall have a latency below 50 ms. It has to be underlined that network latency is only a small part of the end-to-end latency. Another important part is computation and rendering [86].

Others recommend restructuring and simplifying the messaging mechanism while developing a more efficient queuing system and using high speed database engines at the gaming server. The messaging solution lies in handling data messaging with a single packet filled with both game data and mathematically redundant prior message data, like already used in live-feed video imaging [85].

On discussing with game developers [87] the latency and bandwidth are the main topic criteria in gaming. On increasing the bandwidth you will be able to play e.g. virtual soccer world games over the internet which is possible today only in local areas. VR or 3D games seem not to emerge on the playing market in the future because of the tough hardware requirements and the corresponding high costs. An interesting way undertaken by Nvidia [88], a graphic chip supplier: it will deliver a special driver for their graphic chips with 3-D Eyeglasses. This should change a 2-D into a 3-D game on the client without reprogramming it.

6.3 Interactive games for mobile terminals

There are several approaches for the interactive games that need very different requirements. This section implies the case where all players have one copy of the game running on their respective computers, and they interchange only their position into the game and their actions (movements). The data rate is about 1 kbps, the delay must be less than 250 ms, and no FER is allowed. This is a real-time service.

Currently only a very limited set of mobile games offer multiplayer network support. Examples include games like chess, four in a row and use messaging services for the inter client communication. As a consequence, the number of concurrent players is often limited to two players. Mobile phones like the Nokia N-Gage [89] offers multiplayer support over Bluetooth, 2G or 3G networks for multiple (i.e. >=2) players. Portable gaming consoles like the Nintendo DS [90] or PlayStation Portable [91] support communication between consoles or a connection to the internet using 802.11 wireless technologies.

6.4 Summary table of Online Virtual Environments

Table 6-1 summarises the requirements for the online virtual environment services.

Table 6-1: Summary for online virtual environment services

Service	Bit rate	Delay	Jitter	Packet Loss	Mobility	Traffic Priority	Security
MMOG	Relaxed, 56 kbps	< 100 ms (best quality), up to 1s	<10 ms	Highly dependent on the engine from 3% to 35%	No	High	No
Online Distributed Environments	Up to 400 kbps, more in the future	< 100 ms (best quality), up to 1s Future: < 50 ms	<10 ms	Highly dependent on the engine from 3% to 35%	Yes	High	No
Interactive games (mobile)	1 kbps	250 ms	<10 ms	None	Yes	Low	No

7 Remote Technical Services

7.1 Remote Residential Backup

The end-user is becoming himself a content producer rather than a content consumer. This is largely due to the fact that digital video equipments (e.g. camcorders and cameras) are now commonly used even on mobile phones. The files created are then shared with the user friends and relations but most of them are simply stored on a local computer or media centre. These can occupy several tens of gigabytes and some media centres today offer storage space in excess of one Terabyte even for residential use (see for instance the HP MediaSmart Server [92]). It is then becoming more and more important to have a way of safeguarding those files (wedding or children pictures etc…) against destruction (fire, computer fault, accidental deletion, etc…). Remote file backup is consequently a service of increasing popularity (see for instance the Online Backup Vault [93]). However, if the delay requirements for this service are not stringent, the most important performance bottleneck to offer a comfortable service is the network bandwidth. One way around this requirement is to operate the backup during a time when the network is not congested (for instance between 12 am and 4 am in small businesses areas or even between 10 am and 4 pm in residential areas or simply when the computer is idle) and to perform an incremental backup (only the files which have changed are transferred) in order to minimize the quantity of data to copy (although checking and comparing the last modified date of thousands of files can be time consuming as well). A network useful capacity of 1 Gbps would allow the transfer of 1 TeraByte of data in about 2 hours and a half. Finally, the local computers and remote servers must be able to read/write data on to/from their hard drives at a high speed. Today, data transfer rate in the region of 500 Mbyte/s are achievable, fixing the highest necessary speed of the underlying transmission network to achieve a comfortable and practical remote storage/backup solution for residential users.

7.2 Remote home monitoring

With more and more applications and devices in the home with embedded micro processors, it motivates the possibility of monitoring and managing these devices remotely.

Remote Home Monitoring allows the users to monitor and control the home environment conditions (e.g. inside temperature, humidity, etc), security system (sensor status, surveillance camera video or snapshots) and the operational status of a wide range of other home devices and services (refrigerator, freezer, washing machine, light sources, etc…), when away from the home. This can take place through the use of terminals as a PC, PDA, or mobile phone.

However, it is not expected that they will all use a generic interface, and some kind of middleware in the home network is expected. Such functionalities can also be integrated into the residential gateway. The user will be able to access these functionalities from outside the home. The schemes for the user authentication and identification will need to combine convenience with a high level of security. For instance, if a Subscriber Identity Module (SIM)-enabled terminal can be used, then a SIM-based authentication will provide the necessary security. An additional option can be a username and password, or other authentication schemes. Also, different users may have different privileges with regard to the monitoring and control of the home environment.

The requirements for home monitoring vary depending on the expected level information exchange, but in general comparable with those of Internet browsing (for general services like the temperature adjustment) and video streaming (for video surveillance). The key requirement is however the secure access to the devices inside the home and security transmission of data.

7.3 Network Watchdog

The ability for instance to remotely configure end-users devices or terminals from the operator's central office, to diagnose and monitor the network up to the end user, to change/upgrade the firmware installed in the end-user devices is one of the most important issues to ensure end-to-end QoS. The technical report TR-069 of the DSL-forum provides the necessary application layer protocol to enable this [94]. Using this protocol, we can envisage the possibility that the end-users contract some services for some equipment (router, computer, set top box…) to the operator, and when they connect the device at home, it is auto configured according to the signed services. Of course, another benefit is the possibility of manage/repair problems occurring in the end-users' networks or in one of the connected terminals remotely, without the necessity of sending any specialized personnel. It is even possible to manage the access to the applications and version updates.

The requirements might differ remarkably depending on the use. The data rate maybe from 20 kbps when there is command interchange to 10 Mbps if there is some traffic measurement and up to 100 Mbps for heavy computer repair (download of new operating system etc…). The initial delay is less than 10 seconds, and the error is PER <2%.

7.4 Thin Client Applications (remote computer)

In thin client computing, applications are executed on a remote server instead of on the user's device. All user events (e.g. keystrokes and mouse clicks) are sent to the remote server. The server processes the events and generates the resulting screen updates. The graphical data is sent to the client by means of a thin client protocol such as Windows RDP, Citrix ICA and VNC.

Today, quite a lot of enterprises already use thin client computing to deliver the applications to their employees. In the near future, the shift from decentralized computing where every employee has his own PC and maybe also a laptop, to a centralized computing architecture will be clear. The advantages of thin client computing for enterprises are already validated by several success stories. This could be of course valid for residential users in the future considering the multiplication of PCs and connected devices in homes. Similarly, mobile devices such as smartphones or PDAs, have always bigger screen sizes and higher resolutions. Therefore, we can admit that bringing thin client computing functionality on handheld devices is realistic. A big advantage of mobile thin client computing is the reduced power consumption of the handheld device, which is extremely important for mobile users. As such, battery lifetime of mobile devices can be extended when applications are no longer executed on the device itself but on servers in the network.

Thin clients offer many advantages. As all applications are centrally managed by server administrators, the end user is released from configuration troubles and virus definition updates. It also releases the user from regular software and hardware updates. This is key to the further introduction (and multiplication) of computers in homes where the updating/managing/servicing of the customer's computer can be made from the outside for instance by its service provider.

The problem with current thin client protocols is the hampering presentation of multimedia applications [60]. However, recently a hybrid thin client protocol has been developed that allows executing all kind of applications, including multimedia and 3D-applications, with high quality [95].

The requirements for thin client computing depend on the type of application envisaged [96]. Three classes can be distinguished: office applications (e.g. text editor, e-mail program, etc.), browsing the internet and multimedia applications (e.g. video, 3D-game, etc.). However, for all applications the delay constraint is the most important. Indeed, the reaction on a user event can only be seen on the screen after at least one RTT. To guarantee high responsiveness of the application, the total delay must be kept within the predefined requirements of that type of application.

	Example	Bit-Rate	Delay
Type 1	Office-applications	VBR (average 100-500 kbps)	Max 150 ms
Type 2	Browsing the internet	VBR (average 2-4 Mbps)	Max 150 ms
Type 3	Multimedia applications	VBR-CBR (average 4-6 Mbps)	Max 80 ms

Since jitter introduces extra delay, the amount of jitter allowed depends on the network delay and the total allowed delay. Therefore, jitter should be avoided as much as possible.

The requirements for packet loss in the downstream direction (i.e. from server to client) depend on the thin client protocol itself. When the thin client protocol sends incremental updates to the client, it is not acceptable to lose packets, since this would result in unpredictable results shown on the client's device. When the updates are not incremental, small packet loss would be acceptable. Typically, type 1 applications use incremental updates and thus do not allow any packet loss, while type 2 and type 3 applications do not use incremental updates and small packet loss is thus allowed. Packet loss in the upstream direction (i.e. from client to server), for example a keystroke or mouse-click, is not tolerated. Indeed, it would be very annoying for the user to push a button twice before he can see the result on the screen.

Data security is a key driver in the adoption of thin client computing. Users might be concerned about the privacy of their data and might insist upon security warranties. It has to be noted that, only a small piece of hardware is required on the thin client side. The energy consumption of a thin client is about 20 times less than the energy consumption of a PC. For home users, this means a reduction of their energy bill.

7.5 Robotic assistant

Nowadays, some of the most extended robotic assistants are the vacuum cleaners, pool cleaners, or the automatic lawn mowers. It can be expected that in the future, there will be robotic assistants at home to help with the many different every day tasks such as: entertainment (interactive games, play media (audio/video)), surveillance (kids supervision, automatic monitoring of the house (intruder alarm, temperature control)), assistance (helping disabled or elderly people stay at home (pills reminder, emergency calls, reading news).

On the entertainment and monitoring robots, there are many different options on the market. One of the pioneers was Aibo from Sony [97]. The main tasks for these robots were entertaining and making the humans familiar to live with robots around them.

In the future, these robots might have a bigger autonomy of movements and actions and they will be used not only for entertainment, they might be able to provide many other services like VoIP, surveillance and so on. Some of these examples are the Spykee from Mecano [98], the Rovio from WowWee [99] or ConnectR from iRobot (derived from Roomba platform) [100]. This new robot generation has the possibility of being used as development platforms, where the end-users can incorporate their own services. At present it is very hard to determine what network requirements will emerge from this robotic service as the applications can be very diverse.

7.6 Location based services

Location Based Services deliver geographic information between mobile and/or static users via the Internet and/or wireless network. In essence, they provide the capability to find the geographical location of the user (via a mobile device which the user carries) and then provide services based on this location information. Location based services are already very popular nowadays. The best known example is GPS based navigation systems including local information on nearby restaurants, tourist attractions, roadwork and traffic jams etc… such as the "Sprint Navigation" service [101]. Other commercially available services include "Verizon Chaperone" which uses a child phone's GPS to help pinpoint its location [102]. The location information is made available online or on the parent phone. Another example is NTT DoCoMo i-Area which provides location-based services for corporate or residential users [103]. It offers services such as security alert, finding your loved ones or even finding your own mobile phone!

In the future, new indoor services may emerge like ambient control, transfer of multimedia content to the place where the user moves, tracing employees in large office buildings with desk sharing or guiding visitors inside museums, airports, etc... GPS is the prime technology to supply such services. However, it does not work inside buildings due to the absence of a line of sight connection with the satellites. Moreover, these services will often need a very accurate location of the user, more than can offer a GPS system. Other systems available e.g. signal strength based WiFi system may also not be accurate enough. Other complementary technologies like Radio Frequency IDentification (RFID), Ultra Wide Band (UWB) or Near Field Communication (NFC) will be needed.

The transmission of the position information is a real time service, but the network requirements are not very high. Updating the location of a user only needs a very small amount of data. The required bandwidth is of course depending on the update frequency (e.g. 1 Hz), the number of detection devices and the number of devices that are tracked but will be normally not more than 10-20 kbps. When an external server processes the information of the different detection devices, delays certainly should be smaller than the update frequency as otherwise wrong positions could be calculated. As update frequencies will be normally not higher than 1-2 Hz, maximum delays are around 400 ms. Packet loss can influence the accuracy of the localization and should be avoided as much as possible.

7.7 Grid computing

A loose definition of Grid computing is that a Grid interconnects different geographically spread heterogeneous resources (computing, storage, instruments, etc.) and lets them work together to solve problems as a single (albeit virtual) big system, providing uniform access to its constituent components. In a grid context, these ‘problems’ are called jobs. A user typically creates a job, and submits it to the grid middleware, which takes care of all the steps needed to execute the job.

A grid application example is the online visualization of a virtual environment [104]. A virtual environment is typically made up of various objects, described by their shape, size, location, the applied texture, etc. Usually, the description of a scene can be realized in a limited storage space, typically around a few megabytes. Rendering the scene however, requires substantial computational power: a demand of 300 million polygons per second results in required capacities as large as 10,000 GFlops. With a frame rate of 25 fps, latency below 40 ms per frame is required. Assuming a 2.5 MB scene description size (transmission time of 8 ms on a 2.5 Gbps link), about 30 ms is available for processing a frame on the remote resource.

Grids are a good way to minimize energy consumption since their capability of serving high computational and storage demands perfectly fits a thin client scenario (described in Section 7.4), allowing for replacing PCs with less power hungry and longer living client machines delegating jobs to the Grid servers. Though, minimization can be further explored by considering a power-saving scheme effectively reducing energy consumption [105].

7.8 Summary table of Remote Technical Services

Table 7-1 summarises the requirements for the remote technical services.

Table 7-1: Summary for remote technical services

Service	Bit rate	Delay	Jitter	Packet Loss	Mobility	Traffic Priority	Security
Residential Backup	500 Mbps	Relaxed specification	< 10 ms	None (BER<10^{-8})	Yes	Low	No
Home Monitoring	0.1-1 Mbps (surveillance video)	< 400 ms	<50 ms	< 1%	Yes	Low	Yes
Network Watchdog	20 kbps to 100 Mbps	Relaxed specification	< 10 ms	< 2%	Yes	Low	No
Thin Clients	100 kbps to 6 Mbps	< 150 ms (< 80 ms for video)	< 10 ms	None or very small (< 0.1% recommended)	Yes	High	Yes
Robotic Assistant	This is a futuristic service and its requirements are not clear at this time.						
Location Based services	10 – 20 kbps	< 400 ms	< 50 ms	< 3% < 0.1% recommended	Yes	Low	No
Grid Computing	High (1Gbps)	< 40 ms	< 1 ms		No	Variable	No

8 Health / Monitoring services

This section covers the health related services that can be brought to the end user via the broadband access and mobile networks. These services are described as e-health services which are defined as covering all interactions and exchange of data via Information Communication Technologies (ICT) between patients and health service providers or between health institutions covering the access to health information records, telemedicine services and personal connected wearable devices for monitoring and/or supporting patients [106]. The importance of this field is growing rapidly in Europe because of the ageing population and the fact that the cost of individual's healthcare increases as a person gets older. As well, healthcare is a huge market with an average spending of 9% of the GDP for OECD countries [106]. Furthermore, e-health services are seen as a great opportunity to reduce the CO_2 emissions in developed countries by reducing the travels of patients between their home, the doctor's office and the hospital. In a recent report [107], FTTH Council of Europe computed that those services could save around about 65 million kilometres to the patients around Europe.

8.1 Electronic Health Card (EHC) system

Today, in many countries, patients use an electronic card which serves as an identification document to enter databases on which their medical record is kept. Independent databases exist at different locations, for instance, at their general practitioner unit, at the hospital or even at their insurance provider. Each database is independently constructed and is maintained locally. Clearly, the whole system would benefit enormously from a mutualisation and centralisation of the data. A recent survey from IPSOS shows that, in Europe, 85% of the population as well as 85% of all doctors are in favour of having an electronic medical record file that can be remotely accessed by a hospital or a doctor in case of need [108].

Here, we describe an e-health service whereby the patient or doctors can access from any location a unified medical personal record or Electronic Health Card Database (EHC-DB). It is designed to improve the doctors' access to their patients' medical history thus helping them in the diagnosis and treatment processes. On the other hand, it provides patients with the ability of constant overview of their own personal medical data, as well as offering them a variety of additional services (tele-medical services or diagnosis, arranging appointments, etc). The EHC-DB is a place where users' personal and professional accounts are created and managed. It also holds an index enabling quick access to medical data stored locally from any terminal for verified users. The key feature is to provide fast and secure flows of information between various and geographically dispersed medical units. The EHC system will integrate and link all medical units. Below is a description of the different elements composing the whole EHC system:

- EHC Data Base: The data base stores the patients' personal data, its ID and some information about the patients' history of medical services. It serves as an index for dates, places, kinds of services, etc. It does not store patients' medical data (which are stored locally) but manages the access to it. It should keep track of the history of data access. Different levels of data access are granted to various groups of users (patient, medical staff, administrative staff) in order to solve the inherent security issues resulting from centralising such sensitive data.
- Medical units' local Data Bases network: Local DBs should be interconnected and connected to EHC-DB with the EHC-DB working as a coordinating unit. They should work under one standardized data storage and exchange format (set by government units). These are maintained and managed locally.
- Patient's Electronic Health Card (EHC): This will serve as patient's passkey to his personal and medical data. It should contain code enabling access to patient's profile in data base. It should of course be protected with several security solutions. It probably will be issued by national health service provider (e.g. NFZ in Poland, NHS in the UK etc…)

- Medical Staff Electronic Health Card (EHC): This will serve as passkey to the system for Medical Staff. It should be protected in a similar way to the patients EHC. It will as well be issued by the local national health service provider.

The services offered to the patients can be: viewing and verifying his personal data on-line, viewing his medical data on-line, using the card as insurance document (home and abroad), registering for remote medical diagnosis or localisation services (teleEKG, teleEHC, localisation device, etc), arranging appointments at medical centre or physician's practice on-line, etc…

The services offered to the Medical Staff can be: viewing and verifying their patients' personal data on-line, viewing and adding to patients' medical data, checking patients' medical history (globally or in chosen fields), finding and learning about past cases, for help in current diagnosis or for educational purposes (doctor can see only medical data, not patients personal data), inviting colleagues for consultation, etc…

The EHC service system requirements are:

	Data rate	Delay	Traffic priority	Packet loss
Between Medical Units DB	VBR (100 Mbps – 1 Gbps)	up to 80 ms	Guaranteed service	low
Between Medical Units DB and EHC system DB	VBR (100 Mbps – 1 Gbps)	up to 80 ms	Guaranteed service	low
Medical Staff remote access to the system	VBR (2 – 5 Mbps)	up to 150 ms	Guaranteed service	low
Patient remote access to the system	ABR (0,5 – 1 Mbps)	best effort	best effort	medium

Furthermore, jitter should be avoided as much as possible. User events need to be handled as soon as possible to keep the highest level of interactivity as possible. Since jitter introduces extra delay, the amount of jitter allowed depends on the network delay and the total allowed delay. Additionally, it has to be stressed that users of the system should be strictly verified, transferred data should be sent by encrypted data tunnels and integrity of the data should be checked. Finally, the system should accept mobile access for remote users such as ambulances, rescue teams or remote access from patients.

8.2 Tele-Electrocardiogram (ECG)

An electrocardiogram records the electrical activity of the heart over a period of time. Its prime use is to measure and diagnose abnormal rhythms of the heart and identify damaged heart muscle. The goal here is to provide a service enabling the patient to have its his heart activity continuously measured and monitored over time so that quick and effective alert and action can be taken in case of problems while allowing freedom of movement. This system was first used during the visit of pope John Paul II in Poland in 1996 where pilgrims were given the portable monitors and the tele-medical data was transferred via cellular ORANGE POLAND network to a centralised medical centre. A description of Tele-ECG equipment can be found at http://www.pro-plus.pl. The patient wears a portable lightweight device with electrodes attached to his body. This device can transmit the recorded data via the cellular network to a remote medical server called "Cardio SCP server". A tool called "Cardio SCP client" is used by the medical staff to remotely access the server and check, verify, monitor the recorded patient's data.

The requirements on the patient's side are a bit rate over 32 kbps, a short delay and low jitter (regarded as Low Latency Data Service Class) and a medium Packet loss. Of course mobility should be provided along with secure transmission of the data. The requirements for server side are a bit rate over 500 kbps to be able to collect data from ~1000 patients plus a 2 Mbps download (to physician) and 1 Mbps upload (from physician) capacity. Delay and jitter requirements are derived from the Low Latency Data Service Class while medium packet loss can be tolerated. Similar security, mobility and priority requirements as above are necessary.

8.3 Tele-Electrohysterogram (EHG)

An Electrohysterogram is used to continuously monitor the contraction activities of the womb during pregnancy by registering electrical impulses. True contractions of the womb (which occur during labour or just before) are monitored and differentiated from fake ones (which stress unnecessarily the pregnant woman) [109]. Again the goal here is to provide a monitoring service to a patient while maintaining its ability to move around and/or continue to live a "normal" life. Today, such service is ready to be launched on a large scale.

The Tele-EHG monitor device requires similar functionality as the Tele-ECG system, that is the provision of non-invasive sensors for sticking to the skin and a Data Collecting Module. Measurement results are transmitted via a standard cellular phone line or xDSL line with Bluetooth transmission between the sensors. The data is analysed remotely and if an alert is raised then the doctors can react depending on the alert level: send in an ambulance to bring the patient to the hospital, send a nurse to the patient's house for further examination, etc…

The requirements on the patient's side are a bit rate over 32 kbps, a short delay and low jitter (regarded as Low Latency Data Service Class) and a medium Packet loss. Of course mobility should be provided along with secure transmission of the data. The requirements for server side are a bit rate over 500 kbps to be able collect data from ~1000 patients plus a 2 Mbps download (to physician) and 1 Mbps upload (from physician) capacity. Delay and jitter requirements are derived from the Low Latency Data Service Class while medium packet loss can be tolerated. Similar security, mobility and priority requirements as above are necessary.

8.4 Localisation devices (medicine)

The general idea behind the localisation service is to enable the patient to be localised at any time. The localisation information is continuously transmitted to a Remote Service Centre which performs dedicated service functions depending on the patient illness and location. For instance, a patient suffering from Alzheimer disease can be found in case he wanders off and gets lost. In France, it is estimated that 860,000 people suffer from Alzheimer's disease and 225,000 new cases are found each year. Such a monitoring service is predominantly used for patients suffering from Alzheimer's disease but more generally, it can be employed for all diseases affecting patient personality and psychology such as dementia and different other psychic diseases.

In common usage, the localisation device is expected to be equipped with GPS receiver to obtain the localisation information and with mobile technology based transmitter to send the information to the service centre. Localization receiver and information transmitter could vary on technology. Depending on expected localization precision required by the service localization could be obtained not only by GPS, but as well by any localization service proper to 2G/3G technology. Technology used to send the localization information in addition to be a typical GSM transmitter could be as well a WiFi device, if area of service availability is limited (e.g. area of service limited to the hospital).

In France, Orange is proposing such a localisation service [110] together with Medical Mobile [111] based on their localisation bracelet called "Columba". Note that this bracelet is also distributed in 30 countries around the world. The bracelet designed to be used by the patient consist of GPS receiver and GSM transmitter. This is completed by a server/computer system to perform treatment of localization information concerning the monitored person. System performs the verification function, consisting in comparing the current person localization to predefined area. In case the person stays out of the expected area, the alarm is generated. The alarm information as well as location of the patient is therefore delivered to patient family or caregiver.

A very similar approach is developed to monitor patients suffering from diabetes where the glucose content of the blood is recorded on a PDA and sent over to a doctor for survey and, if needed, for intervention [112]. The same scheme can even be applied for less dramatic causes such as diets. [113]

The requirements on the patient's side are a bit rate over 16 kbps, a short delay and low jitter (regarded as Low Latency Data Service Class) and a medium Packet loss. Of course mobility should be provided along with secure transmission of the data. The requirements for server side are a bit rate over 250 kbps to be able collect data from ~1000 patients plus a 2 Mbps download (to physician) and

1 Mbps upload (from physician) capacity. Delay and jitter requirements are derived from the Low Latency Data Service Class while medium packet loss can be tolerated. Similar security, mobility and priority requirements as above are necessary.

8.5 Summary table of Remote Health Services

Table 8-1 summarises the requirements for the remote health services.

Table 8-1: Summary for remote health services

Service	Bit rate	Delay	Jitter	Packet Loss	Mobility	Traffic Priority	Secu rity
TeleEHC	0.5 – 5 Mbps (patient) 100 Mbps to 1 Gbps (hospital/doctor)	80 – 100 ms	<10 ms	None (BER<10^{-8})	Yes	High	Yes
TeleECG	32 kbps (user) 500 kbps – 2 Mbps (central)	80 – 100 ms	<10 ms	None (BER<10^{-8})	Yes (patient) No (central)	High	Yes
TeleEHG	32 kbps (user) 500 kbps – 2 Mbps (central)	80 – 100 ms	<10 ms	None (BER<10^{-8})	Yes, but limited (patient) No (central)	High	Yes
Localisation services (medicine)	16 kbps (user) 250 kbps – 2 Mbps (central)	80 – 100 ms	<10 ms	None (BER<10^{-8})	Yes (patient) No (central)	High	Yes

9 Conclusions

In this document, we have listed the main services either already available to the end-users or still emerging and quantified their requirements for the networks transporting them (access, mobile and in-building). In order to be able to compare the requirements of these services, a common metric to evaluate their needs is defined. These parameters are summarized below:

- Bit-rate/Throughput
- (Tolerable) Delay
- (Tolerable) Jitter
- (Tolerable) Packet loss
- Mobility
- Traffic Priority
- Security (required or not)

The first four parameters (Bit-rate to Packet Loss) are measurable quantities that can translate directly into network requirements and specifications while the last three (Mobility to Security) are non-measurable and serve at defining the context of the specific service usage.

Six services classes are identified each corresponding to a usage group and similar types requirements:

- Basic communication such as instant messaging, e-mail, and telephony.
- Internet-related services such as general browsing, e-banking, e-shopping and similar (including file sharing using peer-to-peer)
- Video-related services such as Video on Demand, IPTV and video conferencing
- Online Virtual Environments such as a social network or gaming
- Remote Technical services such as the ability to remotely control/survey your home
- Remote Health services such as remote health monitoring

For each service class, the underlying services and applications have been listed. Firstly, we have concentrated on the services that are already available to the end-users and how we can forecast their evolution in time. Then, we have provided additional information regarding the future emerging services (for instance, the immersive TV). If the network requirements of current services can be precisely identified, those of future applications are also easier to estimate. A summary of the services requirements for each class can be found in sections 3.4, 4.5, 5.9, 6.4, 7.8 and 8.5.

The main conclusion from this service enumeration/grouping is that the different networks will transport services that have an ever increasing need for bandwidth and more and more stringent requirements in terms of delay as services become more and more interactive and video based. As well, it has to be noted that the current trend shows that the services that were once confined to a particular type of network can now be transported by a variety of networks (a user may want to check his emails from its desktop computer but also from its mobile phone or PDA). This implies seamless handovers as well as unified service management requirements.