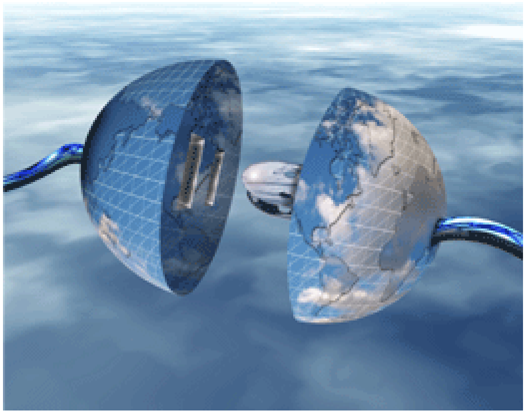
Multimedia Delivery in the Future Internet

A Converged Network Perspective



Media Delivery Platforms Cluster

White Paper

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1 INTRODUCTION

The term “**Networked Media**” implies that ***all kinds of media including text, image, 3D graphics, audio and video are produced, distributed, shared, managed and consumed on-line through various networks, like the Internet, Fiber, WiFi, WiMAX, GPRS, 3G and so on, in a convergent manner*** [1]. This white paper is the contribution of the Media Delivery Platform (MDP) cluster and aims to cover the ***Networked*** challenges of the *Networked Media* in the transition to the Future of the Internet.

Internet has evolved and changed the way we work and live. End users of the Internet have been confronted with a bewildering range of media, services and applications and of technological innovations concerning media formats, wireless networks, terminal types and capabilities. And there is little evidence that the pace of this innovation is slowing. Today, over one billion of users access the Internet on regular basis, more than 100 million users have downloaded at least one (multi)media file and over 47 millions of them do so regularly, searching in more than 160 Exabytes[1] of content. In the near future these numbers are expected to exponentially rise. It is expected that the Internet content will be increased by at least a factor of 6, rising to more than 990 Exabytes before 2012, fuelled mainly by the users themselves. Moreover, it is envisaged that in a near- to mid-term future, the Internet will provide the means to share and distribute (new) multimedia content and services with superior quality and striking flexibility, in a trusted and personalized way, improving citizens’ quality of life, working conditions, edutainment and safety.

In this evolving environment, new transport protocols, new multimedia encoding schemes, cross-layer in-the network adaptation, machine-to-machine communication (including RFIDs), rich 3D content as well as community networks and the use of peer-to-peer (P2P) overlays are expected to generate new models of interaction and cooperation, and be able to support enhanced perceived quality-of-experience (PQoE) and innovative applications “on the move”, like virtual collaboration environments, personalised services/ media, virtual sport groups, on-line gaming, edutainment. In this context, the interaction with content combined with interactive/multimedia search capabilities across distributed repositories, opportunistic P2P networks and the dynamic adaptation to the characteristics of diverse mobile terminals are expected to contribute towards such a vision.

Based on work that has taken place in a number of EC co-funded projects, in Framework Program 6 (FP6) and Framework Program 7 (FP7), a group of experts and technology visionaries have voluntarily contributed in this white paper aiming to describe the status, the state-of-the art, the challenges and the way ahead in the area of Content Aware media delivery platforms.

[1] 1 Exabyte = 2^{60} Bytes

2 MARKET ENVIRONMENT & BUSINESS MOTIVATIONS

Internet has evolved and changed the way we work and live. Moreover, the population of the Internet end users is growing and becoming quite diverse. It typically comprises corporate employees in the office or whilst travelling, along with consumers in residence at home or whilst on holiday. The user expects no longer to be exclusively connected to the Internet through a computer, but also through his personal digital assistant, cell-phone, television or any other type of Internet enabled device.

In this chapter, we'll try to highlight the current and Future Internet market environment, the business motivations and opportunities and the social impact of the new generation of the Internet.

2.1 Social Impact and User Services

The evolving Internet has changed the typical user main service needs. Much innovation on services has centred on creating change in the way we work, live and for life in general (e.g. e-mail, text messaging, e-commerce, gaming and search tools etc). In the Future Internet, new services will be designed, constructed and marketed in ways that are highly adapted to current and future human needs (e.g. the need to socialise, for new sensations and entertainment, for ease of use anywhere and anytime, to mitigate travelling and save time and energy etc). The new services may be summarized in the following group of services:

User Connectivity provides people with continuous connectivity with the Internet (e.g. home, whilst travelling, work, play) through a variety of devices. This will provide people at any time and place with:

- Capability to easily exchange/download their own choice of media e.g. music, film, video selections;
- New ways to express themselves to others on the Internet through media;
- New ways to socialise with each other through media via communication (e.g. Skype, Powwownow), social networking (e.g. Myspace, Facebook, Bebo, Friendster, LinkedIn) and on-line diaries (e.g. Blogs, Wikis).
- New ways to capture the attention of millions of viewers via content sharing sites (e.g. YouTube, Flickr, Upcoming, del.icio.us, Last.fm, and 43 Things);
- New ways to interact with things (so called "the Internet of Things") where short-range mobile transceivers are embedded into a wide array of additional gadgets and everyday items, enabling new forms of communication between people and things (smart presence), and between things themselves (smart space); tele-presence users communicate with each other and with devices from remote locations e.g. to control household environment (heat, lights, TV).
- Seamless services to users over heterogeneous wireless networks; Users when roaming between coverage areas of mobile, broadcast terrestrial/satellite and wireless LAN/MAN/PAN require uninterrupted service during the transition e.g. a seamless transition.
- Inter-connectivity and inter-working between multitudes of devices that can be used by a single user. In today's device-centric world with its need to configure each device separately is contrary to the desired ease of use.
- New types of mobile services such as mobile multi-media mail, phones as portable entertainment players, mobile marketing for retailers and manufacturers, multi-channel shopping, navigation, phones used to acquire tickets and money, and mobile intranet applications.

Other services may include, but definitely not limited to ***higher video quality*** (High Definition, Digital Cinema Definition and Ultra High Definition video), ***3D views*** (including Stereo, Multiview, Holoscopy, Holography), ***Gaming***, ***User Participation and User Communities, new Search Tools, Internet Advertising***.

2.2 Vision of the Future Internet

The key differences between the current and the Future Internet may be summarized in the following:

True Broadband, which will provide Gigabits (or even higher) per second connectivity, will enable a wider range of higher quality services such as 3D Video and Graphics in wired and wireless networks.

Sharing of Control: Whereas the first generation Internet facilitated the sharing of information (e.g. media streaming, web, email, search engines, interactive applications etc.), the second generation Internet will facilitate the sharing of control in order to manage the provisioning of services through universal usability (user focus: real-time adaption of form-factors, formats, security, quality, mobility protocols; personalization of user interface, services, interest profiles, interactivity and user inclusion), ubiquitous computing (hardware focus: interoperation of disparate devices) and distributed control (software focus: distributed and controlled processing for across networks) which will used to provide solutions for security, mobility, reliability, resilience, availability, problem analysis, networking, protocol flexibility, connectivity, scalability and quality of service.

Increasingly the Internet is becoming a mission critical tool both at home and in the work place.

- **At Home:** At home the Internet is being used more and more for every day tasks such as purchasing (e.g. cinema tickets, travel tickets, weekly shopping), finances (e.g. banking, tax returns, share management), information gathering (e.g. children's and adult's homework, holiday destinations) and communication (e.g. with distant relatives and work place colleagues). Furthermore as the cost of oil increases due to its scarcity, forcing people to increasingly work more and more from home, the Internet is becoming increasingly more important for that growing band of people who predominantly work from home or are self employed and work from home. It is thus a critical tool for the post fossil fuel era.
- **In Workplace:** In the corporate organisation work place the Internet is also being used for every day tasks such as purchasing, product support, finances, information gathering, marketing (e.g. reaching a worldwide audience, sales network management), communication (e.g. between colleagues in a large and international organisation, recruitment) media product delivery (e.g. delivering media products such as TV programs to satellite, terrestrial and cable access points). When the network is "down", due to network or power outages, the workplace has become so dependant on the Internet that the workings of the whole organisation grinds to a near halt as the main means of communication is lost.
- **Transition Strategy:** Having recognised that the Internet is a mission critical tool for the economics of the home, organisations and the country as a whole, suitable transition/evolution strategies should be established towards introducing the Future Internet.

The Future Internet may be considered from the service provider, the network provider and the mobile network provider perspective:

2.2.1 Service Provider Perspective

The Future Internet will be able to help Service providers offer their customers the benefits of new tools that will respond to user needs and become more user-centric and enable natural communication between people residing in distant locations without incurring the cost and trouble of travelling:

- **New Media Sensations**: 3D media through multi-layered/multi-viewed content coding, considering the evolving H.264 AVC/ SVC/MVC and their emerging successors, as the major foreseen A/V coding technologies for multimedia content distribution over heterogeneous networks/terminals and large audiences. Virtual 3D collaborative platforms create new requirements in terms of information representation, filtering, aggregation and networking so as to provide real-time 3D navigation combined with physical and emotional involvement of the user. New 3D content formats from efficient mixing of real 3D captured content with Computer Generated Graphics (CGG) able to offer new visual sensations to the users.

- **Search Tools:** New network information manipulators and algorithms for an efficient 2D/3D content search (including search in complex virtual environments). Increasing demand towards more sophisticated (multimedia) search tools, including tools for media professionals and P2P overlay networks.

2.2.2 Network Provider Perspective

- **Ability to choose price/performance:** The Future Internet will be able to help Network providers offer their customers the benefits of new tools that will facilitate the sharing of media, the establishment of media communities whilst providing the ability to choose price/performance that the customer wants based around Reliability, Resilience, Availability and QoS.
- **Increased Capacity:** As we move from ADSL (with 12 Mbit/s downstream and 3.5 Mbit/s upstream) to VDSL (with 100 Mbit/s duplex at a range of about 500 meters), or even Fiber to the Home (FTTH) more different types and quality of streamed and communication services can be delivered.
- **Reliability, Resilience, Availability:** The Future Internet will provide reliability, resilience and availability through network tools that will provide:
    - *Network awareness* which will allow the network to cooperatively detect and mitigate for network faults and congestion by dynamically changing the route or code rate of media codecs (3D, video from mobile to Ultra-HDTV).
    - *Media awareness* which will allow the network to provide different error correction measures depending on the type of media that is being transmitted and the QoS required by the user.
    - *Joint media and network awareness* (e.g. media-to-network cross-layer dynamic adaptation using network aware video coding techniques) to improve video quality beyond HDTV, towards 3D and Ultra HDTV whilst providing reliability, resilience and availability of services.
    - *Autonomic network awareness* that will provide automatic problem analysis tools which can self-heal the Internet in the case of congestion, link or node faults.
- **Security and Access Control:** Network owners define their own security policies and will do so also in the future. Some organizations, like military may require closed high security networks with highly hierarchical policy structures, while it is reasonable to assume that public institutions will provide more open networks. Therefore the Future Internet will be able to provide a wider range of different security products by network operators. Furthermore security systems created for one system may need to be interoperable with other security systems (or at least interface).
- **Trust:** Media owners will define their own DRM, content sharing (identity management), ownership and trading of personal and professional virtual digital objects, right of use policies for their different content.
- **User Connectivity:** The Internet of 3D Media will be characterised by an increasing demand for personalisation through the aggregation of services, supported by high throughput multimedia streams and sessions including data from smart objects and reduce start-up/modification/adjustment delays and increasing interactivity to support real-time multi-party network sessions, supporting virtual 3D worlds for professional as well as community and gaming applications.
- **Networking:** The Future Internet will offer balanced network requirements for: Content distribution, Distributed control, Caching, P2P multi-source/multi-network content streaming.
- **Quality of Service:** In order to involve media industry in network and services provisioning, the Future Internet will be able to provide network architecture with a range of different price/performance points for delivering a range of different quality services with the ability to provide QoS data from the edges of the autonomous systems within the Internet.

2.2.3 Mobile Network Provider Perspective

- **Increased Mobile Capacity:** offered by new generation of converged mobile networks will provide improved mobile services and applications. The first applications that take advantage of these higher rates will emerge (multimedia presentations, telemedicine services, mobile teleconference, interactive entertainment, high-quality music download, etc);
- **Mobility:** The Future Internet should support user mobility where the user is allowed access to any network environment, terminal mobility where the terminal is able to operate in any network environment, network mobility where one (radio) network is able to be connected to any other network and service mobility where a service is accessible on any terminal over any network. This will provide businesses with the opportunity to provide different types of mobility services to people on the move;
- **Mobile Protocols:** The Future Internet should support protocols that support mobility (e.g. for seamless handover of AV services and for the support of mobile sensors efficient transfer of small data units).
- **Scalable Services:** A Future Internet that provided scalable user Interface and scalable 2D/3D codecs would permit mobile devices to access wired Internet sites.
- **Programmable/Configurable Terminals:** Programmable/configurable mobile terminals for the future Internet would allow both complimentary and competing radio transceivers and different conditional access systems to coexist on a single phone. Scalable codec and media adaptation methods are required to support the huge variability of mobile phone types in the market.
- **Improved Addressing:** resulting in efficient access to caches, proxies and the support for mobility.

2.2.4 User Perspective

The increased pace of technological innovation in recent years in the field of networks and digital media is contributing to a more user-centric approach in the future development of the Internet. Fused by their artistic hunger and creative aspirations, users of this new era are becoming increasingly involved beyond the consumption, in the creation of digital media (photos, graphics, video, 3D content, etc). The Internet has been one of the key thrusts in this user-authored content revolution, which would have a significant contribution and impact in the TV and Film world:

- **Collaborative authoring:** Connectivity would enable users to author collectively a piece of content such as a video to produce a new form of TV and film publishing. In this new form collaborative online tools would facilitate the manipulation and collaborative editing of content by several users at remote places in a both professional and amateur (user-generated) context.
- **Interactive Content Enrichment:** User-authored content would become a new content source for broadcasters and other service providers, enriching their regular programming and encouraging the creative and artistic aspirations of the new generation who wishes to share content with the rest of viewers. In this users would be able modify and enrich content at both the end-user terminal (household) and head-end (service provider) site.

3 MULTIMEDIA CONTENT IN THE FUTURE INTERNET

The last couple of years, a number of new standards have evolved in the area of (rich) media content format. Yet, one of the first issues that will differentiate multimedia delivery in the Future Internet is the format of the multimedia content. In this chapter, we review the main existing and enabling multimedia content formats and standards. From the various multimedia content formats, we mainly focus on the video formats, as video streaming is nowadays the most demanding multimedia application over the Internet.

3.1 Video coding state of the art

One of the greatest issues related to media delivery platforms is scalability: the ability to transmit the same multimedia stream to multiple heterogeneous terminals and each one to receive the piece of information that can, aims, is allowed to get (layered multicast). Various video coding standards (MPEG-2 Video/H.261, H.263, and MPEG-4 Visual) already include some scalability tools. However, their scalable profiles have rarely been used, due to their significant loss in coding efficiency as well as an increase in decoder complexity.

By the end of 2007, the **Scalable Video Coding (SVC)** activity of Moving Picture Expert Group (MPEG) of ISO/IEC and Video Coding Expert Group (VCEG) of ITU-T will have been finished their joint standard as H.264 Annex G and MPEG-4 Part 10 Advanced Video Coding Amendment 3. This standard is expected to outperform all existing scalable video coding solutions in terms of coding efficiency and required computational power, ensured by a complete single-loop decoding process.

The SVC extension is built on H.264/MPEG-4 AVC and re-uses most of its innovative components [7]. As a distinctive feature, SVC generates an H.264 / MPEG-4 AVC compliant, i.e., backwards-compatible base layer and one or several enhancement layer(s). The base layer bit stream corresponds to a minimum quality, frame rate, and resolution (e.g., QCIF video), and the enhancement layer bit streams represent the same video at gradually increased quality and/or increased resolution (e.g., CIF) and/or increased frame rate.

In general, parts of a scalable bit stream can be decoded with reduced quality (temporal resolution, spatial resolution or SNR). The updates from one quality (in one of the scalable directions) to the next higher quality can be seen as data elements in a data cube model (Figure 1). For scalable video there are temporal, spatial and SNR levels (see the three dimensions in Figure 1). A scalability level includes the bits for exactly one quality step in exactly one direction. The SVC elementary stream itself contains data in fully scalable representation, which means that decoding is possible at any operation point given by the scalability levels in a three-dimensional space, as shown in Figure 1.

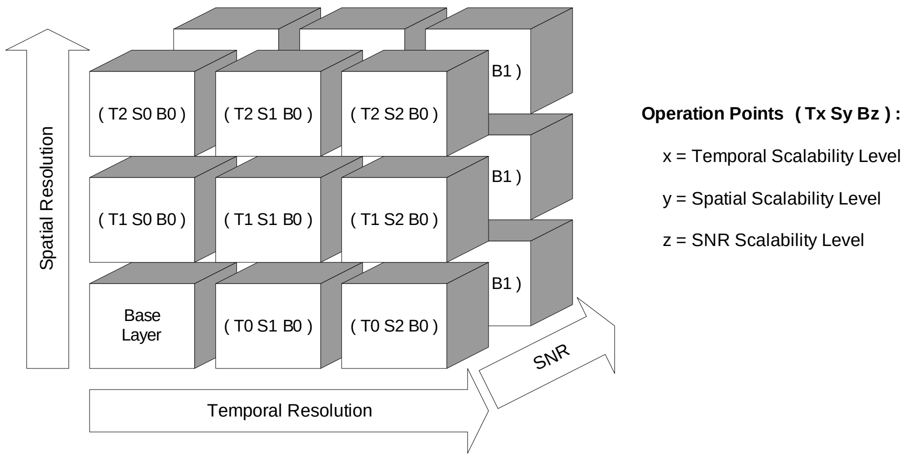

Figure 1: Data Cube Model [7]

Alternatively a bit stream can be organized in so-called *Tiers*. A Tier contains all scalability levels to update the video from one quality to the next; it must enhance the quality in at least one direction (temporal, spatial or SNR). This representation offers simple adaptation operations at defined qualities by discarding unneeded layers. Figure 2 shows an example of a scalable bit stream organized in four layers. In this example, the base layer (layer 0) contains the video at QCIF@6fps. Layer 1 contains the updates to CIF@12fps. Layer 2 enhances the SNR of layer 1 and updates to CIF@24fps, the full temporal resolution. Layer 3, finally, contains the updates for obtaining 4CIF spatial resolution. These mappings of the data elements are defined a priori depending on the requirements imposed by an application or by a user or service.

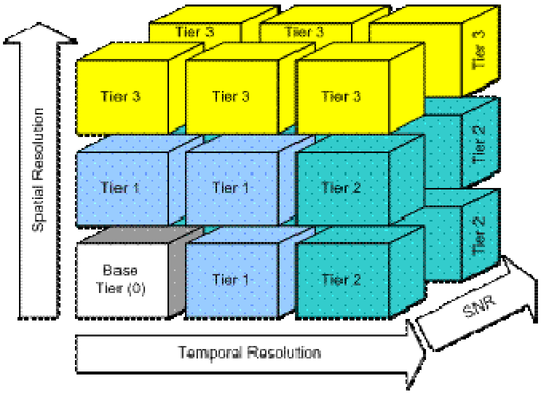

Figure 2: Scalable Bit Stream Representation using Tiers [7]

The application space of SVC is very versatile. In communication scenarios for instance, client devices with different display sizes or computational resources can be jointly addressed by a single stream. The SVC stream can also be adapted to the current congestion of the access network. These features are important for video conferencing and mobile broadcast. An example of the generated SVC stream is shown in Figure 3. An SVC Encoder generates an SVC stream from a live camera (i.e., a football match) containing several resolutions (e.g., QCIF, CIF, SD, HD), frame rates (e.g., 15, 25, 30 fps) and bitrates (e.g., 500, 1000, 2000, 4000, 12000 kbps). These configurations would allow cell phones, PDAs and HDTV receivers to decode and present the video content.

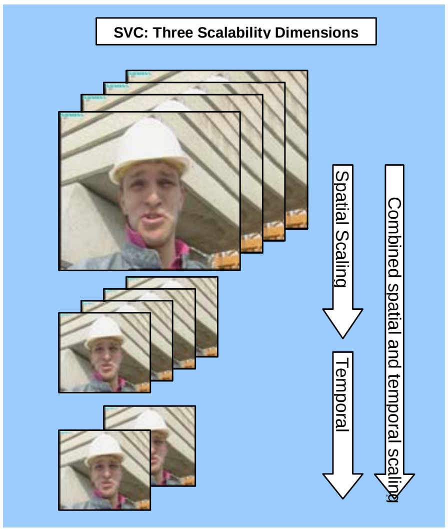

Figure 3: Scalability dimensions in SVC

3.2 Future Multimedia Content Formats Challenges

Future Internet poses a number of challenges related to the multimedia content types and format.

3.2.1 High Definition Content

While HD may seen as a reality today, it has been observed that many products and formats are not yet compatible: “Full HD”, “HDMI”, … Decides the increased resolution, when it is considered from the end-user perspective, HD is a quality enhancement, which offers increased PQoS.

3.2.2 3D Content & 3D Content Creation

Many different approaches have been adopted in attempts to realise free viewing 3D displays. Several groups have demonstrated autostereoscopic 3D displays, which work on the principle of presenting multiple images to the viewer by use of temporal or spatial multiplexing of several discrete viewpoints to the eyes. However, these autostereoscopic 3D displays are not truly spatial displays since they exclude vertical *parallax* and rely upon the brain to fuse the two disparate images to create the 3D sensation. As a result stereo systems tend to cause eye strain, fatigue and headaches after prolonged viewing as users are required to focus to the screen plane but converge their eyes to a point in space, producing unnatural viewing. With recent advances in digital technology, some human factors which result in eye fatigue have been eliminated. However, some intrinsic eye fatigue factors will always exist in stereoscopic 3D technology.

Creating a truly realistic 3D real-time viewing experience in an ergonomic and cost effective manner is a fundamental engineering challenge. Future 3D technology should seek to advance the current existing technologies not only in capturing and manipulating 3D content but also in creating a new 3D content format which offers fatigue free viewing with more than one person independently of the viewer's position. 3D holoscopic and holography are two technologies that overcome the shortcomings of stereoscopic imaging, but their adoptions for 3D TV and 3D cinema are still in their infancy. Holographic recording requires coherent light but offer the ultimate 3D viewing experience. Holoscopic video uses microlens arrays to recording a 3D scene and can operate under incoherent illumination, which is in contrast with holography, and hence it allows more conventional live capture and display procedures to be adopted. Future 3D video could use different technologies for 3D content creation and display.

Content creators always look for new forms and ways for improving their content and adding new sensations to the viewer experience. There has been a trend in cinema in producing films with 3D enriched content such the latest animated adventure film "Beowulf". However, novel forms of 3D content, should also find its way into small and medium size content creation companies, moving the experience from cinema halls and cinema projectors to the everyday household environments and computers, providing increased number of audiences with a taste of the versatility and power of 3D as both consumers and producers. There is a need to make 3D easier to capture, edit and share over the Internet and more available to small production houses and everyday people

Hence an important challenge in 3D content creation and presentation is to provide the user similar hardware and software facilities as those enjoyed today by 2D video makers and users. To that effect it is important that the 3D content can be captured using a single 3D camera and displayed on flat panel 3D display, and it is also important to provide software tools for creating 3D virtual content, mixing, sharing, editing and searching.

Key topics that will influence the 3D technology of the future are:

- 3D content that can interest consumers and attract them to 3D video
- 3D content distribution technology and formats that will bring this content to the consumer's home
- 3D display technology capable of producing HDTV-quality images for the consumer in both 2D and 3D modes.
- Accessibility to software and hardware tools which will allow 3D content creation, including live-action and computer graphics, and 3D content manipulation.
- 3D content compatible with the large existing 2D content creation and distribution infrastructure.

3.2.3 Multiview Video Coding (MVC)

Recent technology advances significantly extend the sensation of classical 2D video. Those new types of applications allow the user to freely choose a viewpoint of a visual scene or/and provide a 3D depth impression of a visual scene.

Figure 4 shows a Free viewpoint TV (FTV) system. As it is shown, FTV can generate very natural free viewpoint images in real time. A scene is captured using a dense array of synchronized cameras. The camera images are placed in a simple manner forming a Ray Space [8] that allows rendering the scene from any position by simple processing. So the user can view the scene not only from the original camera positions but also from any virtual viewpoint. FTV supports a wide variety of applications since it can be applied to any kind of scene.

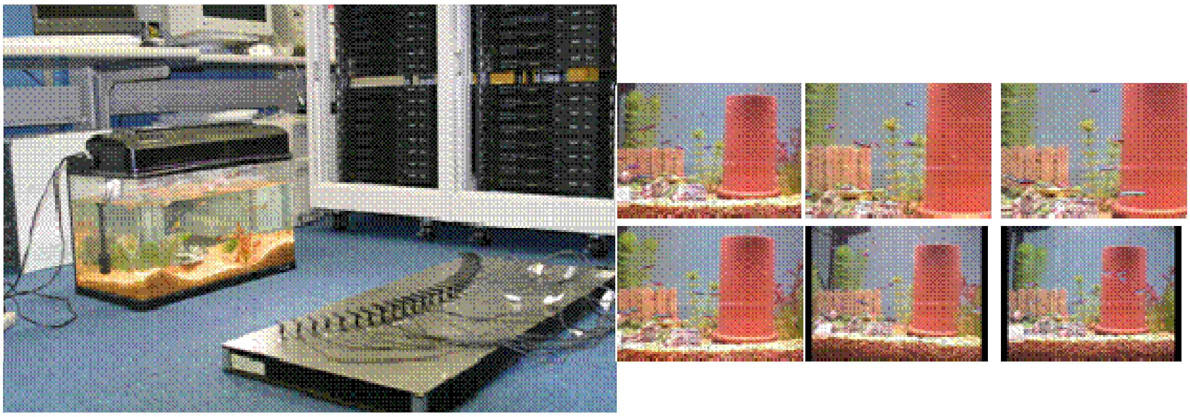

Figure 4: Capturing and processing parts of FTV system. & Free Viewpoint Images generated [10]

FTV and FVV (free viewpoint video) are interesting for user applications (DVD of an opera/concert where the user can freely choose the viewpoint) as well as for (post-)production. Systems for the latter are already being used (e.g. for sports, movies, EyeVision, Matrix-effects).

The second new functionality provided by these new technologies is a 3D depth impression of the observed scene. In fact, this functionality, also known as stereo, is not new. Extending visual sensation to the 3rd dimension has been investigated for a long time. Commercial systems (e.g. in IMAX theatres) are available. However, acceptance for large user mass markets (3D-TV at home, DVDs, etc.) has not been reached yet. This may be overcome due to recent developments of 3D displays (where no more glasses are needed) and advanced 3D rendering that supports head motion parallax viewing.

Figure 5 shows an example of a 3D-TV system. A scene is again captured by N synchronized cameras [10]. The multiple video signals are encoded and transmitted. At the receiver they are decoded, rendered and displayed on a 3D display. 3D rendering means creating 2 views, one for each eye, which if perceived by a human will create a depth impression. There are several types of 3D displays available, with and without glasses, and therefore also different types of specific 3D rendering algorithms.

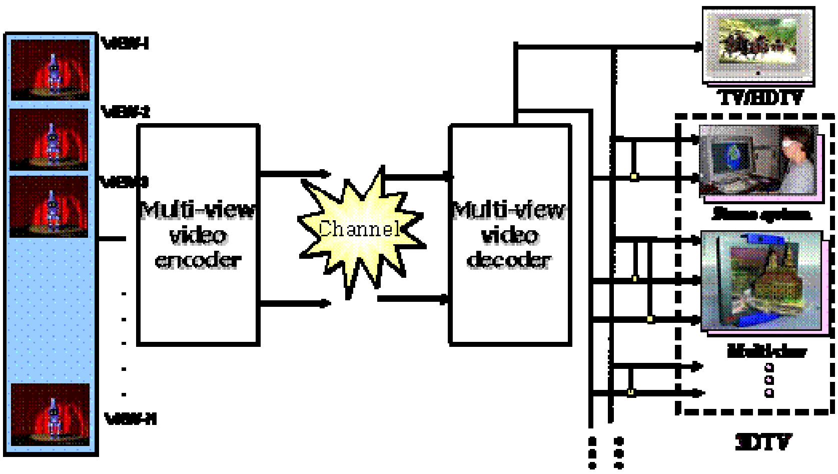

Figure 5: Example of a 3D-TV system [10]

A common element of all systems described above is the use of multiple views of the same scene that have to be transmitted to the user. The straight-forward solution for this would be to encode all the video signals independently using a state-of-the-art video codec such as AVC. However, specific multi-view video coding (MVC) algorithms give significantly better results. The main goal of MVC is therefore a compression gain for multi-view video.

MVC is a common building block for various systems as described above, for efficient compression of multi-view video data. However, more work will be necessary to realize standards conforming FTV/FVV/3D-TV systems.

3.2.4 H.265

H.265 is a long-term video coding standard, launched by the Video Coding Expert Group (VCEG) of the ITU-T. As indicated in ITU-T SG16's homepage [8], "VCEG plans to complete the requirement definition and begin the detailed algorithm design for H.265 in the study period 2005-2008, and if the progress in contribution technology is sufficient, H.265 is expected to be finalized in 2009-2010". Despite the initial plans, H.265 hasn't been formalized yet, and VCEG keeps seeking proposals and information regarding the possibility of a major gain in performance to justify the step from H.264 to H.265. Though the necessary scope of H.265 is yet to be determined, it is agreed that among the goals will be:

- simplicity and "back to basics" approach
- high coding efficiency, e.g., two times compared with H.264
- computational efficiency, considering both encoder and decoder
- loss/error robustness
- network friendliness
- other considerations as necessary

Backward/forward compatibility is not assumed to be required for H.265, as H.265 is a brand new standard instead of an extension of H.264.

During the recent 3-4 years, the contributions to VCEG mainly focus on improving coding efficiency. To better evaluate these contributions and retain progress, key technical areas are developed as the software platform, which uses JM11 as the baseline and continuously integrates promising coding tools. Sources say that H265 should be 50% more efficient than H.264. However, there is no yet evidence of readiness of technical advances sufficient to justify embarking on a concentrated effort toward H.265, and all the contributions are actually in the direction of an "H.264+".

3.2.5 MPEG/Laser

As of today there are few solutions for Rich Media standard browser, which cover both broadcast and telco environments. One of them is the MPEG4 part 20 "LASeR". ISO/IEC 14496-20 defines LASeR (Lightweight Application Scene Representation) and SAF (Simple Aggregation Format) designed for devices needing low footprint rich media browsers. LASeR is a binary encoding of SVG Tiny 1.2 with dynamic updates added, and SAF is a very simple configuration of the MPEG-4 Sync Layer, allowing aggregation and synchronization of elementary streams, secure streaming on HTTP, and streaming with RTP through the mapping defines in RFC 3640. The requirements for LASeR and SAF include footprint, code size and performance in addition to the usual MPEG criteria of compression, so as to cater with the specific needs of constrained devices such as mobile phones. LASeR and SAF allow rich-media mobile services to be designed and streamed to a majority of today's phones with the current network capabilities and protocols. The specification will be final as soon as SVGT1.2 goes final.

LASeR and SAF have been proposed as work items to OMA and 3GPP. A 3GPP work item called DIMS (Dynamic interactive Multi-media Scene) based on a subset of LASeR is currently finalized and it is expected to be adopted soon. OMA has adopted a rich-media environment (RME) work item, which is based on a subset of LASeR, will be a superset of DIMS.

4 CONVERGED NETWORKS

The next step, towards Future Internet Media Delivery Platforms will be a converged network architecture, able to offer truly broadband access anywhere and anytime. In this chapter, we present various converged wireless and fixed-wireless networks. Focus has been mainly put at the access networks, since they have been for years the major obstacle to interactive multimedia distribution.

4.1 State of play in Network Technologies

The vision of broadband access form anyplace at anytime is becoming a reality due to advances in the fixed and fixed-wireless access and the mobile networks.

4.1.1 Fixed Access Networks

For years the major barrier to a “digital networked house” has been the access network. The inadequate network infrastructure and the huge cost of new installations had formed the well-known “last mile problem”, which hindered the broadband access at home. Connecting each house to broadband access networks however, represents an unprecedented opportunity to set the stage for a vast range of new home applications and added value services to residential users and expand the customer base beyond the corporate environment.

In the near future, a large portion of customers will be connected to the information superhighway via optical fibers. However, at least for the next decade, several economic and geographical reasons will still prevent the introduction of dedicated optical fibers to the majority of customers. Instead evolutions of existing access network technologies, such as Asymmetric Digital Subscriber Line (ADSL), Hybrid Fiber-Coaxial (HFC), and Passive Optical Networks (PON) are expected to be employed in the physical layer. These technologies are already available to a significant percentage of the developed countries population.

4.1.2 Fixed-Wireless Access & Mobile Networks

The last couple of years, in addition to landline access networks, wireless access networks and mobile networks are becoming increasingly important. In the early days it was the installation cost that turned wireless access into very applying solutions. Today, it is the flexibility and the simplicity.

Prominent examples of fixed-wireless technologies are the WiFi and the WiMAX, which has provided the ability to offer broadband services to stationary users in the home and the last mile without needs to invest in upgrading and extending landline infrastructures. Moreover, DVB-x technologies have become quite important the last couple of years.

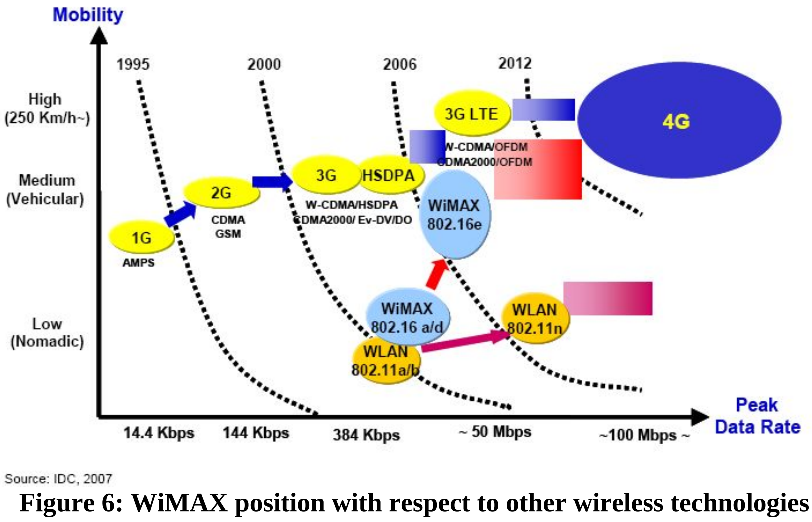

Figure 6: WiMAX position with respect to other wireless technologies

As illustrated by Figure 6, WiMAX is at the convergence between the mobile communications 1G → 3G standards track (depicted in yellow) and the residential wireless WiFi track (depicted in orange), before a yet hypothetical "4G" solution. Supporting point to multi-point (PMP) broadband wireless access rates up to 2+ Mbps, over a minimum coverage area of 3-5 miles, offering a relatively easy deployment due to low network investment costs, and the capability to operate in non-line-of-sight (NLOS) over licensed or non-licensed radio spectrum, WiMAX is undoubtedly an attractive technology. Yet, the market success of WiMAX is far from certain due to a number of reasons including the lack of standardization and interoperability, the lack of agreed spectrum band worldwide for WiMAX and the increasing broadband competition, leading to price compression, that makes some wonder about the interest to introduce a new standard. Finally, it seems that the Mobile WiMAX may not live to be the obvious successor of HSxPA techniques for mobile phone wireless, as the GSM Association (GSMA), which federates mobile operators worldwide, has made known during last 3GSM Asia in November 2007, that she was adopting LTE (Long-Term Evolution) technology and not WiMAX nor Qualcomm ultra mobile broadband (UMB) proposal. The increasing number of complimenting and competing wireless standards is shown in Figure 7.

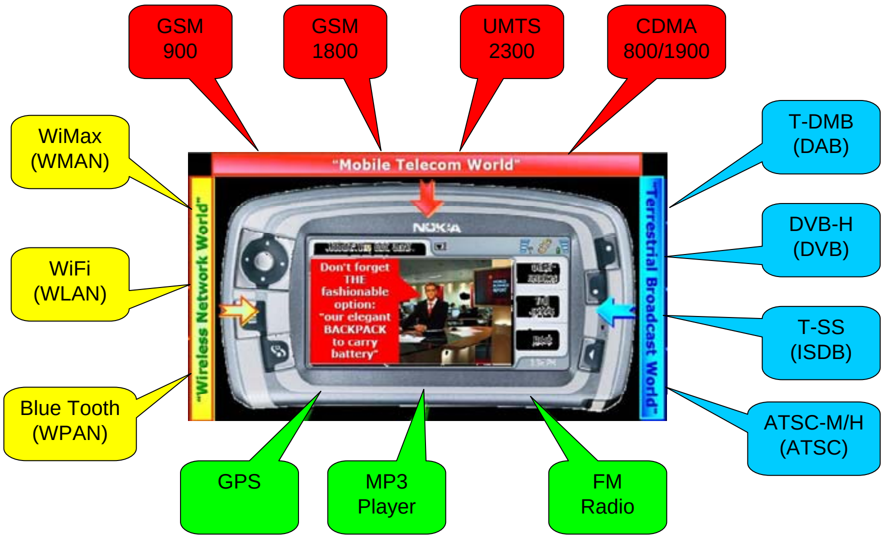

Figure 7: Complimenting and Competing Wireless Standards

4.1.2.1 Long Term Evolution (LTE)

Even though the 3GPP Release 5, 6 and 7 for HSPA have improved UMTS's capabilities significantly, to ensure competitiveness of the UMTS in the long term, 3GPP is currently working on its newer Release. The 3GPP's Release 8 or the so called "Long-Term Evolution" (LTE) has set its target to develop a high-speed, low-latency, low-cost and packet-optimized radio access technology which closely meets the definitions of the so-called 4G mobile network.

Some of the initial targets for the specification effort of LTE include:

- Peak data rate of more than 100 Mbps on the downlink and more than 50 Mbps on the uplink, given available bandwidth of 20 MHz, 2 receiving antennas and 1 transmitting antenna at the User Equipment (UE).
- Average user throughput should be several times better than that of Release 6
- Reduced latency both in control and user planes. A small IP packet for the user plane should also experience latency only around 5 ms in an unloaded condition.
- Scalable system bandwidth deployment of 20, 15, 10, 5 MHz or even less should be supported.
- Inter-working between UTRAN, GERAN and non-3GPP networks. Roaming between other 3GPP's Radio Network Technologies (RAT) must be possible. Also, the system must be able to co-exist with previously deployed GERAN or UTRAN.

- End-to-end QoS should be supported. VoIP services should have comparable quality to circuit-switched voice services of UMTS.

Once completed, there will be 2 main parts in this new architecture. The first is the radio interface part which is evolved from the original UTRAN, hence the name E-UTRAN. The second part is the Evolved Packet Core (EPC), a result from the work item “Service Architecture Evolution (SAE)”, which is an all-IP core network to be discussed in the next section.

4.1.2.2 Multimedia Broadcast Multicast Service (MBMS)

The high demand of the same streaming services by groups of mobile users is outdating the dedicated connections due to the high resources consumption. If the information is to be transmitted to large users’ groups then sharing the network resources is the best option. In broadcast mode, the transmissions occur regardless of user presence in the given broadcast area. In the case of multicast, it may be thought as a unidirectional point-to-multipoint service in which the information is conveyed from a single source to a set of active users inside a multicast area. Therefore, users need to have an uplink channel to perform the service activation. In both cases, the service efficiency increases with the number of users intending to receive the information.

Brought in under R6 of 3GPP, Multimedia Broadcast and Multicast Service (MBMS) is the system used to do broadcast and multicast packet data on UMTS networks to large users’ groups. Hence, MBMS allows the unidirectional transportation of information from a single source to multiple receipts (point-to-multipoint). MBMS makes available two types of user services: *streaming* and *file delivery*. The streaming services provide a stream of continuous media, such as audio and video; the file delivery services are used to deliver file data over an MBMS channel. The MBMS architecture allows an efficient use of the network resources, by sending the “same” information to all MBMS users simultaneously.

The introduction of MBMS into the UMTS network forces the modification of several network elements and, as a result, new interfaces and protocols are defined. Also, a new entity is presented: the Broadcast Multicast Service Centre (BM-SC). BM-SC is the platform responsible for the management of the broadcast and multicast content that is inserted into the UMTS network. It makes available functions for the provision and deliver of MBMS user services.

4.1.2.3 Service Architecture Evolution (SAE)

The SAE is a Work Item of 3GPP to develop a framework for an evolution or migration of the 3GPP system to a higher-data-rate, lower-latency, packet-optimized system that supports multiple Radio Access Technologies (RATs). It is to be an all-IP core network part of the LTE along with being a connection point for other access technologies, both from 3GPP and non-3GPP. This includes most of the mobile network technologies, e.g. HSPA, UMTS, GPRS, WiMAX and also several other landline access networks (e.g. xDSL). The SAE will have a simplified and more cost-effective architecture by minimizing the number of nodes needed in the network. It also aims at providing more effective protocols, support for services with high performance requirements, support for mobility between heterogeneous access networks and to provide lower delays and higher throughput.

The SAE architecture consists of three functional elements:

- **MME (Mobility Management Entity)** It provides UE’s mobility between 3GPP access networks
- **Serving Gateway** which provides packet routing and forwarding functionality
- **Packet Data Network Gateway (PDN-GW)** which provides per-user based packet filtering, UE IP address allocation and serves as a connection point to the Public Data Network including non-3GPP access networks.

Figure 8 shows the functional entities of the SAE and their interconnections with other networks.

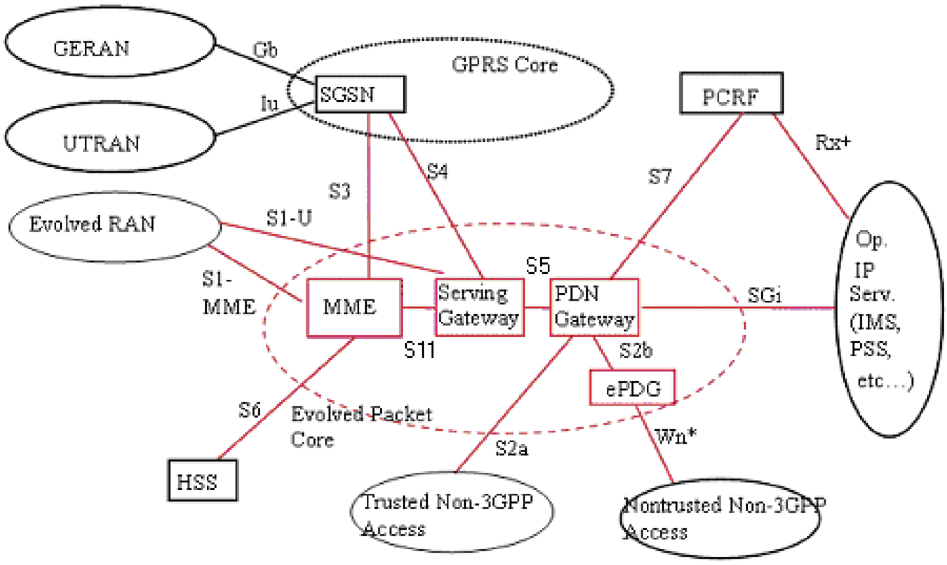

Figure 8: SAE architecture for the non-roaming case

An important point from the SAE architecture is the SGi interface that connects the SAE and any additional network-edge module. The SGi is a reference point between the PDN GW and the packet data network. Packet data network may be an operator external public or private packet data network or an intra operator packet data network, e.g. for provision of IMS services. This reference point corresponds to Gi and Wi functionalities and supports any 3GPP and non-3GPP access systems.

4.1.3 P2P Overlays Architecture

In addition to the access and core networks providing connectivity to the endpoints, most of the Internet content delivery solutions require an infrastructure on top of those networks (overlay) in order to be able to manage users and contents and support the multimedia distribution properly. These overlays can be centralized, partially decentralized or fully decentralized, mainly according to where are their functionalities located: in the network elements and/or in the end-user equipment. The decentralized solutions are commonly known as Peer-to-Peer (P2P). P2P distribution has emerged as a viable solution due to its ease of deployment, low cost of operation and scalability. P2P distribution has also the advantage over client/server systems of offering more resources to clients by effectively turning each one of them into a secondary server that assists in the distribution of the stream from the original server. From a peer's point of view all other participating peers are potential servers to which it can switch if its overlay connection with the currently servicing one degrades. Peer selection is one of the major architectural components of a P2P streaming system affecting performance and viability of the system.

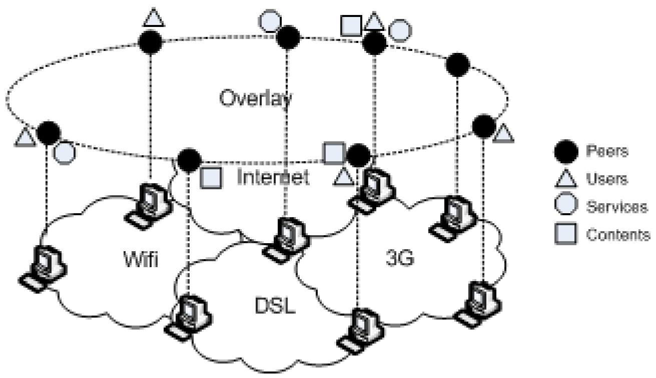

Figure 9: P2P Overlays Architecture

4.2 Future Challenges

The next generation of fixed and fixed-wireless networks have provided broadband access from home and office. The next generation mobile networks, such as EGPRS and UMTS, have also gained much interest in their abilities to offer higher throughputs, lower cost-per-bit of information and their more "friendliness" to IP traffic. New Radio Access Networks (RAN) like the HSPA, LTE and Mobile-WiMAX all have all-IP core networks which, combining with improved scheduling and error-recovery mechanisms in the MAC layer, make them possible alternatives as access technologies to traditional landline networks. This is especially the case when taking into account that some services are used while users are on the move.

Wireless networks, both for mobile and stationary users, are still prone to some common drawbacks. They are heterogeneous in bandwidth, reliability and receiver device characteristics. In wireless channels, packets can be delayed (due to queuing, propagation, transmission, and processing delays), lost or discarded due to complexity/resource limitations or display capabilities of the receiver. Hence, the experienced packet losses can be up to 10% or more, and the time allocated to the various users and the resulting goodput for multimedia bit stream transmission can also vary significantly over time.

This variability of wireless resources has considerable consequences for multimedia applications and often leads to unsatisfactory user experience as they often do not match the following requirements;

- **High bandwidth**—many consumer applications, for example, High-Definition TV, require transmission bit rates of several Mbps.
- **Very stringent delay constraints**—delays of less than 200 ms are required for interactive applications, such as videoconferencing and surveillance, while for multimedia streaming applications, delays of 1–5 s are tolerable. Packets that arrive after their display time are discarded at the receiver side or, at best, would be used for concealing subsequently received multimedia packets.

Fortunately, multimedia applications can cope with a certain amount of packet losses depending on the used sequence characteristics, compression schemes, error protection and error concealment strategies available at the receiver (e.g., packet losses up to 5% or more can be tolerated at times). Consequently, unlike file transfers, real-time multimedia applications do not require a complete insulation from packet losses, but rather require the application layer to cooperate with the lower layers to select the optimal wireless transmission strategy that maximizes the multimedia performance.

Figure 10 gives a roadmap of future research challenges towards Integrated Content Service Infrastructure by approximately year 2025 [9]. These are focused on content networks and services for communities and home users, including Wireless Mesh Networks, P2P technologies, media streaming, QoS, content descriptions, and transcoding. They provide the foundations for the first generation of CNs for community networks that integrate CDN services, delivery and content management.

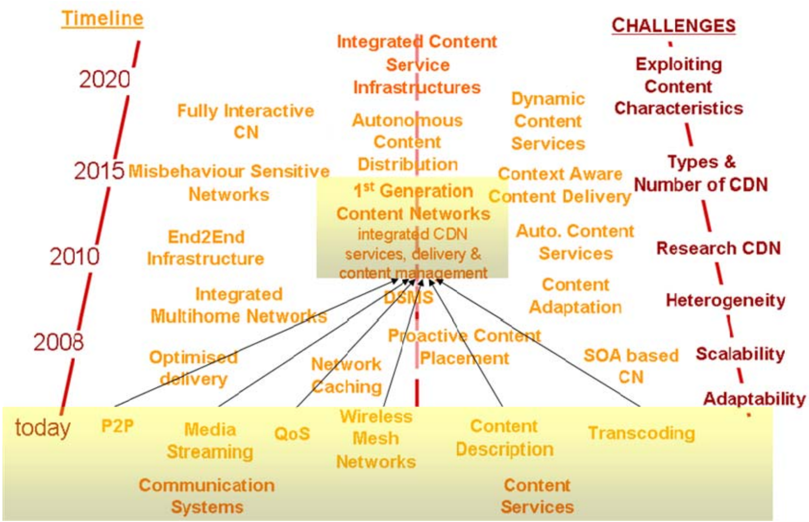

Figure 10: Long-term and short-term roadmap

4.2.1 Wireless Mesh Networks

Wireless mesh networks (WMNs) are comprised of a backbone of mesh routers which collect and relay the traffic generated by mesh clients. Mesh routers have limited (if any) mobility and are usually connected through wireless links. Mesh clients are typically mobile and rely on mesh routers to deliver data to the intended destinations. The use of wireless links causes communications among routers to suffer from environmental noise and interference problems. To provide connectivity, new MAC protocols have been developed, which enable nodes to switch their radio to a different channel when needed. However, the channel switching requires fine-grained synchronization among nodes in order to avoid the deafness problem, i.e., the transmitter and the intended receiver may be on different channels. Also the time for channel switching which can be in the range of a few milliseconds to a few hundred microseconds may be unacceptable for most real-time multimedia applications.

Recently, given the availability of low cost wireless devices, a different solution for the problem of reducing the interference is being proposed, which consists in endowing each node with multiple radios. Radios are set to different channels and no channel switching is required. Thus, each node can simultaneously communicate over different channels, which has been shown to reduce the interference and increase the network throughput. Challenging research issues in multi-radio wireless mesh networks are the channel assignment and the routing problems, i.e., the problems to find, respectively, an assignment of channels to radios and a set of flow rates for every network link which optimize a given objective. Common optimization objectives are to maximize the aggregate network throughput, to verify the achievability of a traffic demand vector or to maximize the minimum end-to-end rate.

Channel assignment and routing are not independent problems, as solving one problem requires a solution for the other problem. Indeed, solving the routing problem requires the knowledge of the bandwidth available on all the links. However, we do not have this knowledge prior to solving the channel assignment problem. This is because the channel assignment algorithm determines the sets of links sharing the same channel and accordingly their available bandwidth. Likewise, the channel assignment algorithm needs to be aware of the flow rate expected on the network links. This information enables the algorithm to assign channels such that the available bandwidth on each link exceeds the required flow rate. However, this information can only be determined by solving the routing problem. Hence, it is clear that the channel assignment and the routing problems are closely inter-dependent on each other and must be jointly solved. Unfortunately, the joint channel assignment and routing problem is NP-complete.

A WMN is typically modelled as in Figure 11. Some of the mesh routers denoted as *mesh aggregation devices* collect user traffic and forward it to the wired network through multiple hops across the WMN. Mesh routers connected to the wired network are denoted as *mesh gateways*.

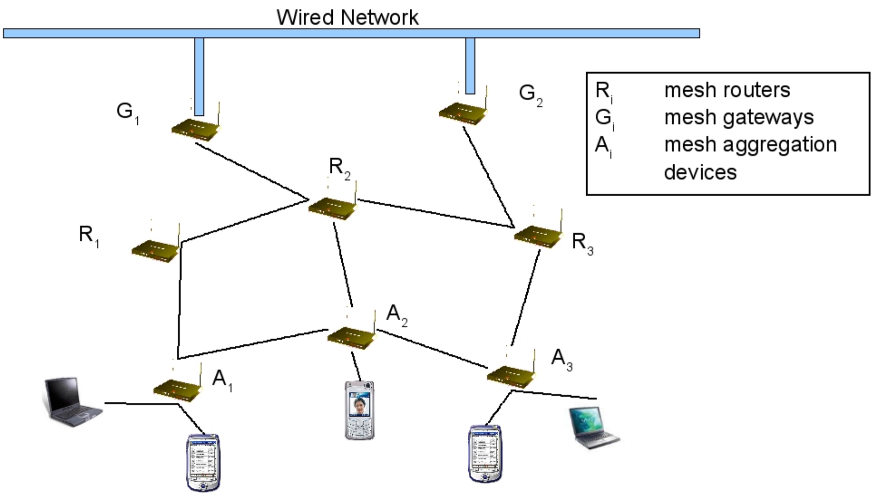

Figure 11. WMN reference architecture

4.2.2 QoS & Resources Management in Wireless Access Networks

Wireless Local Area Networks (WLANs) diffusion is encountering an increasing success due to low deployment costs and high flexibility of such infrastructures for the building of small, medium and wide Wireless Access Networks. Typical architectures proposed by vendors and widely accepted by customers consider Wireless Access Networks composed of two typologies of infrastructure devices: Wireless Termination Points (WTPs) and the Access Controllers (ACs). WTPs provide basic connectivity services, while ACs provide configuration, management and monitoring services. However, more management functions and services can be implemented in the ACs, increasing further the configuration and management flexibility of the architecture. In order to provide interoperability among WTPs and ACs of different manufactures, the IETF is defining a new protocol: the CAPWAP Protocol (*Control And Provisioning of Wireless Access Points).* The goals claimed in the specifications of CAPWAP are:

1. Centralize authentication and policy enforcement functions for a wireless network.
2. Move processing away from the WTP, leaving there only time critical functions.
3. Provide a generic encapsulation and transport mechanism.

Resource optimization in Wireless Access Networks is a critical task. Indeed, the limited resources present in those Access Networks, the high exposition to interferences and the increasing demand of high resource-demanding services (VoIP calls, Video calls and AV-Streaming) request particular attention in the configuration of the network and in its management. Among the others, the research community has identified three problems as the most critical for Wireless Access Networks:

4.2.2.1 Frequency planning

The IEEE 802.11b is the most widely used standard for WLANs. IEEE 802.11b defines eleven transmission channels for wireless communication, but at most three of those channels can be used simultaneously in the same area without cross-interference. When configuring or upgrading large deployments of WTPs, the configuration of frequencies used by every WTP may be a major problem in optimizing overall network performance. Optimal frequencies reusing is desirable, reducing interference among adjacent wireless cells, but it is not always possible. In this context several works modelled and proposed solutions to the optimal allocation of frequencies to WTPs.

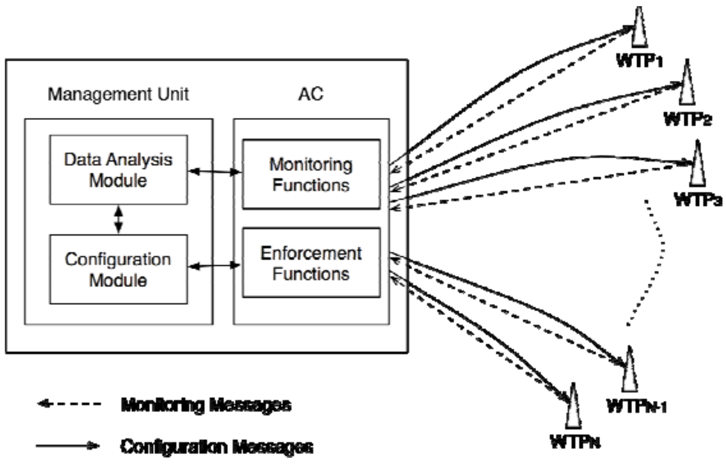

Figure 12: CAPWAP-based architecture for Frequency Planning.

When dealing with huge installations of WTPs both configuration and frequency re-planning upon failures, requires automatic algorithms solving the frequency distribution problem. In this context there is a rich research activity in the definition of such algorithms and especially in algorithms based on CAPWAP functionalities. Figure 12 shows a CAPWAP-based architecture supporting frequency planning and based on automatic configuration/reconfiguration algorithms. In such architecture, each WTP is assumed to monitor the wireless channel checking the activity of other WTPs in neighbour communication channels.

This information, periodically sent through the AC, using CAPWAP monitoring messages, to an optimization module, can be used to decide whether or not the current configuration is satisfactory. When the configuration is not satisfactory, a reconfiguration algorithm can enforce a new frequencies distribution using the enforcement functionalities that CAPWAP provides.

4.2.2.2 Load Balancing

Load Balancing deals with the problem of users distribution among WTPs. Several studies showed that in Wireless Access Networks, users are often unevenly distributed in space and, hence, the number of associated users may vary widely from WTP to WTP. This may translate, since some WTP can result highly congested, in an uneven load distribution, which can severely degrade the quality of services that users communications receive. Load Balancing aims at mitigating this problem by forcing some users to roam toward low loaded (neighbour) WTPs and avoiding congestion.

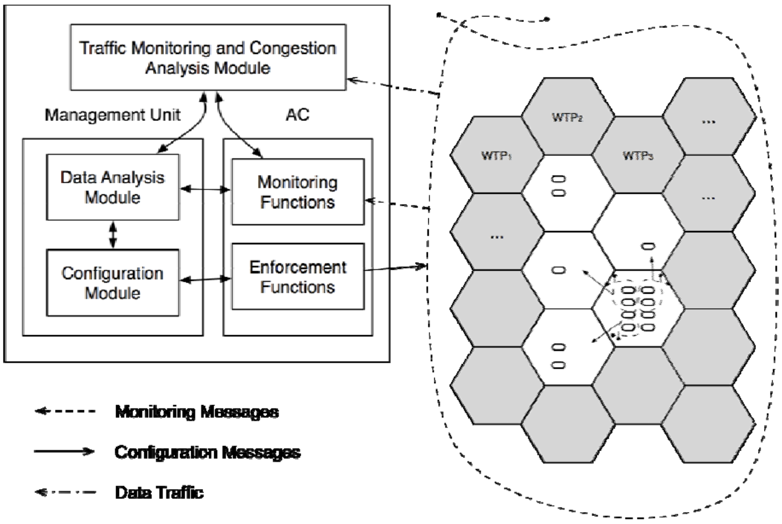

Figure 13: CAPWAP-based Load Balancing Architecture

IEEE 802.11 does not provide any explicit mechanism to remotely control the association of users to a specific WTP. In order to implement Load Balancing strategies, a set of extensions to the standard has been proposed. Cell Breathing gained a lot of consensus due its simple and standard implementation. Cell Breathing is based on the assumption that stations choose to associate with the WTP from which they receive beacons with the highest power. Hence, Cell Breathing tries to redistribute users association varying the power that each WTP uses for transmitting beacons, increasing the power of beacons sent by low loaded WTPs, and reducing that of beacons coming from highly loaded WTPs.

An example of architecture for Load Balancing is shown in Figure 13. The architecture may use CAPWAP functionalities in order to obtain information about WTPs congestion and association of users, monitor channel activity and congestion in neighbour WTPs. A complementary monitoring module can integrate previously introduced functionalities with per flow statistics in order to better measure the quality experienced in each WTP. Eventually a configuration module may vary beacons power in order to implement Cell Breathing and force users to move.

4.2.2.3 Future Scenarios for Resources Management in Wireless Access Networks

Several strategies for resource management and network configuration can be proposed, analyzed and tested. Moreover, the simple analysis of a joint solution to previously presented problems is still missing, opening to a lot of promising research in this field. From a long-term research perspective, we can expect that various management strategies and algorithms, analyzed in mid-term research, will be used together for the definition of autonomic solutions to the management of Wireless Access Networks. In this context we devise several research challenges:

- ***Virtualization management***. Both IEEE 802.11 and CAPWAP support the implementation and management of multiple virtual Wireless Networks, sharing the same physical devices. This introduces a great flexibility in resource allocation, charging modelling and users' management. This flexibility requires some efforts in the definition of proper management strategies, complicated by the interaction within and between Virtual Wireless Access Networks.
- ***Fully automatic managements systems for (Virtual) Wireless Access Networks.*** In this context the number of possible solution is unlimited. Moreover it is still unclear the way the single components solving the mid-term problems described above have to be integrated in semi-automatic solutions for near future networks.
- ***High performing mobility management integrated with Load Balancing and QoS support***. This challenge is particularly interesting due the diffusion of metropolitan public infrastructure for Internet access. Due to mobility, heterogeneity and the number of involved elements, those scenarios are particularly research and industrial challenging, while they are of particular interest for Service Providers.
- ***The impact of misbehaving nodes***. Misbehaving nodes intend to increase their own benefits (such as higher throughput or extended battery life), without regard to the overall functioning of the network. Research has shown that this can lead to severe throughput degradation. The latest drivers and the EDCA mechanism of IEEE 802.11 make it easy for a malicious user to gain an advantage over standard-compliant users [11]. In multihop networks, it is possible for a misbehaving user to lower the priority of forwarded traffic or even (through a firewall) drop the packets of others. All these actions are easy to perform and have shown to give substantial benefits in terms of throughput. However, they mostly require expert skills and should therefore be less common in community networks.

4.2.3 Provision of End-to-End QoS

Effective content network services rely on highly efficient and cost-effective mechanisms that can provide support for (different levels of) end-to-end QoS. In current literature, a number of approaches fall under the set of routing metrics, and the routing algorithms using those metrics within the inter-domain routing protocol, trying to find a path that simultaneously satisfies *n* independent QoS constraints. This Multi-Constrained Path problem is the focus of QoS Routing (QoSR), and it has been proven to be NP-hard when the number of constraints on multiple additive or multiplicative metrics is $n >= 2$.

Therefore, several researchers have contributed with innovative heuristics aiming to find suboptimal solutions to this problem, but with the advantage of being computable in polynomial time. Most of these heuristics fall into the intra-domain routing area, but some years ago the issue also started to become surveyed at the inter-domain level. In this latter case, the complexity increases significantly mainly due to the fact that although a suboptimal path may be found, stringent end-to-end QoS still demands for inter-domain resource reservation. This reservation requirement essentially reveals the connection-oriented nature of QoS, but following such an approach at the inter-domain level, imposes - at least at present - several tough challenges in practical terms. As an alternative, it is possible to conceive dynamic end-to-end inter-domain QoS without any kind of resource reservation, and to follow the IP connectionless paradigm, as long as only soft end-to-end QoS is guaranteed. The main advantage of this kind of approach is that it is indeed much more cost-effective than any inter-domain reservation-based approach, and depending on the frame of the proposal and its implementation, it may also end up being highly efficient.

From this perspective the initial inter-domain heuristics mostly tended to propose QoS and Traffic Engineering (TE) extensions to BGP, but quite recently some research groups and manufacturers have started to avoid new enhancements to the protocol and proposed to decouple part of these tasks from BGP devices. While the initial set of inter-domain heuristics is only able to improve end-to-end performance for internets under low routing dynamics, the latter heuristics end up being much more effective, especially, when routing changes occur more frequently. The main difference between these two approaches is that the latter decouples part of the policy control portion of the routing process from BGP devices. Hence, the two approaches basically differ in how policies are controlled and signaled. In-band QoSR and TE techniques, - that is, those inherently supported by BGP - can feasibly operate over long time-scales, rendering them appropriate for static or pseudo-static QoSR and TE provisioning. On the other hand, out-of-band techniques - that is, those decoupled from BGP - are in fact able to operate at much shorter timescales and, thus, they are very appropriate for dynamic or even highly dynamic QoSR and TE provisioning. However, the stability implications of rearranging inter-domain traffic at very short time scales is not yet understood. Indeed, the effect of managing large amounts of inter-domain traffic in this way is completely unpredictable. Thus, these kinds of solutions are definitively not applicable, for example, to large transit Autonomous Systems (ASs), such as Tier-1 or Tier-2 Internet Service Providers (ISPs). Additionally, the rearrangement of small fractions of inter-domain traffic at short timescales, magnified by the number of sources simultaneously injecting these perturbations into the network, may also end up being unpredictable in terms of overall stability.

Above all, stub multihomed ASs are those which could benefit the most from novel mechanisms, providing them with dynamic QoSR and/or TE capabilities at medium or short timescales. In fact, nearly 80% of the more than 20000 ASs that compose the Internet are stub ASs, where the majority of this fraction is multihomed. Multihoming is a widespread practice exploited by stub ASs, which consists of using multiple external links to connect to different ISPs. By increasing their connectivity to the Internet, stub networks can potentially obtain several benefits, especially, in terms of resilience, cost, and traffic performance. These are potential benefits since multihoming by itself is unable to guarantee the improvement of any of them. Thus, additional mechanisms are needed so as to accomplish such improvements. In particular, when an online mechanism actively controls how the traffic is distributed and routed among the different links connecting a stub network to the Internet, it is referred to as intelligent or smart route control.

Figure 14 presents a scenario of two stub ASs (i.e., AS1 e AS2) employing smart route control. For instance in this figure, the smart route controller (SRC) of AS 2 might improve the performance of the outbound traffic toward the remote stub AS, i.e., AS3, through switching among the paths AS3-ISP1-ISP3 and AS3-ISP1-ISP4-ISP5 across the ISP3 and ISP5, respectively.

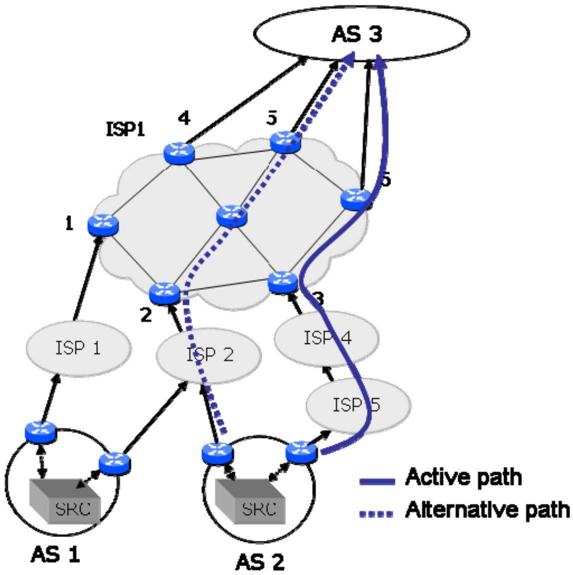

Figure 14: Simple scenario of two Multihomed ASs employing Smart Route Control

This particular fraction of ASs crowds together mostly medium and large Enterprise Customers, Content Service Providers (CSPs), and small Network Service Providers (NSPs), which altogether actually represent around 80% of the total number of ASs in the Internet. Therefore, the blast of stub multihomed ASs in the last few years has gained huge interest in both research and commercial fields. For these ASs the issue of QoSR at the inter-domain level arises as a strong need. Whereas some research groups rely on QoS and Traffic Engineering extensions to BGP, others tend to avoid new enhancements to the protocol and propose Overlay networks to address the subject. While the former approach provides significant improvements for internets under low routing dynamics, the latter ends up being more effective when routing changes occur more frequently. The main idea behind the overlay concept is to decouple part of the policy control portion of the routing process from BGP devices. In this sense, the two approaches differ in how policies are controlled and signalled. BGP enhancements tend to provide in-band signalling, while the overlay approach provides out-of-band signalling.

4.2.3.1 The Overlay Architecture

The Overlay Architecture is mostly appropriate when communicating domains are multihomed, and thus may need some kind of mechanism to rapidly change their traffic behavior depending on network conditions. In fact, multihoming is the trend that most stub ASs exhibit in nowadays Internet, which mainly try to achieve load balance and fault tolerance on the connection to the network. In addition, present inter-domain traffic characteristics reveal that even though an AS will exchange traffic with most of the Internet, only a small number of ASs are responsible for a large fraction of the existing traffic. Moreover, this traffic is mainly exchanged among ASs that are not directly connected; instead they are generally 2 to 4 hops away.

Therefore, the combination of all these features is designed with a focus on QoSR among strategically selected non-peering multihomed ASs. The proposed architecture to inter-domain QoSR is based on a completely distributed Overlay Architecture and a routing layer for dynamic QoS provisioning, while using QoS extensions and TE capabilities of the underlying BGP layer for static QoS provisioning. Within the overlay inter-domain routing structure reside special Overlay Entities (OEs), whose main functionalities are the exchange of Service Level Agreements (SLAs), end-to-end monitoring, and examination of compliance with the SLAs. Thus, in this model, two peering OEs belonging to different ASs spanning across several AS hops are able to exchange an SLA and agree upon a set of QoS variables. The intermediate ASs do not need to participate in the Overlay Architecture, and therefore no OEs are needed within these transit ASs, which is a great advantage when compared to other approaches (such as the QRON approach). To illustrate the model, Figure 15 presents an example of a simple scenario of two peering OEs (i.e., OE S, OE D) located at two remote multihomed stub ASes.

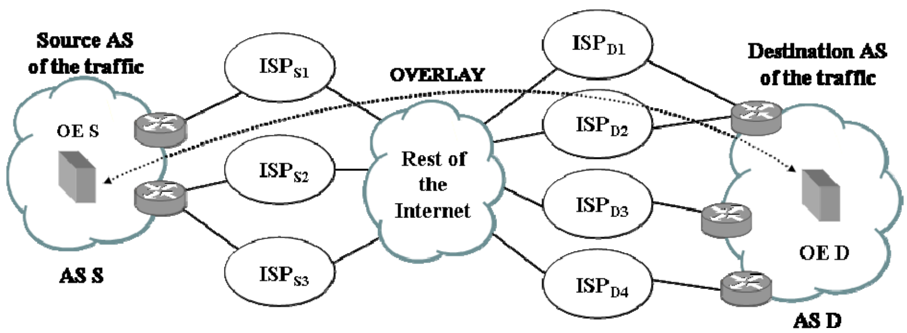

Figure 15: Illustration of a possible scenario of two peering OEs

The real challenge is to develop a completely distributed overlay system, where each OE behaves in a reflective manner. In this sense the overlay approach is like facing a mirror. Instead of proposing a complex scheme to dynamically and accurately manage how traffic enters a target AS, the proposed research tasks will focus on how traffic should exit from the source AS. Hence, the OE within the source AS has to behave like the image in a mirror of the OE in the target AS. This mirroring scheme allows the OE in the source AS to dynamically manage its outgoing traffic to the target AS depending on the network conditions, and the compliance with the previously established SLA for a given set of CoSs. Then, under normal network conditions, and based on an efficient static QoS provisioning, each OE should measure end-to-end QoS variables and check for violations to the SLA for every CoS. Within each multihomed AS participating in the Overlay Architecture, its OE will provide tools to carry out those measurements along every link connecting the AS to the Internet. Here, the time scale needed to detect and react to service level degradation or a link failure is very small when compared with the BGP time scale. The nature of this overlay structure acts as a complementary layer conceived to enhance the performance of the underlying BGP layer containing both QoS and non-QoS aware routers.

4.2.3.2 Next Steps In Signalling (NSIS)

Next Steps In Signalling (NSIS) is a signalling framework being developed by the IETF in the context of the NSIS Working Group, for the purpose of installing and maintaining flow state in the network. Among other uses, NSIS can be used in assisting the provision of end-to-end Quality of Service.

NSIS is based on various signalling protocols, the main one being RSVP. The intention is to reuse RSVP mechanisms whenever possible, since these mechanisms have already been widely tested, leaving out all unnecessary complexity (e.g., multicast support). It is, thus, a simpler and more scalable approach to resource reservation, when compared to RSVP. By using a two-layer signalling architecture, signalling transport is separated from signalling applications, which opens the way to develop several such applications, of which quality of service signalling is the first use case. The initial requirements of NSIS include support for the independence of signalling and network control paradigms, ability to place NSIS initiators, forwarders, and responders anywhere in the network through on-path and off-path signalling, transparent signalling through the network, grouping of signalling for several micro-flows, flow aggregation, scalability, flexibility, and security.

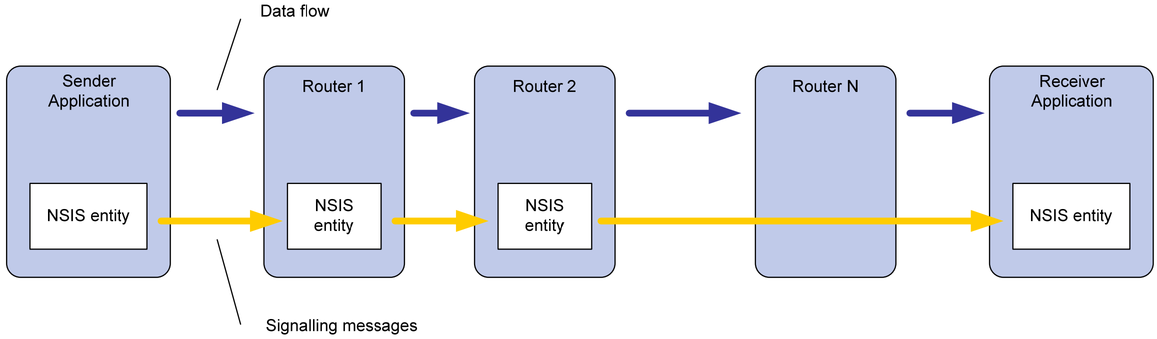

Figure 16: Simple NSIS scenario

Although NSIS can work on a per-flow basis, it allows flow aggregation based on the use of the DSCP field or on tunnels. Additionally, it works on a hop-by-hop basis, between NSIS-aware nodes (NSIS Entities, NE, also referred to as NSIS hops). Nodes not supporting NSIS are transparent, which means that there is no need for deployment of NSIS in every network entity. This is illustrated in Figure 16, where the end-systems and two of the routers support NSIS Entities that exchange signalling messages related to the data flow. NSIS allows signalling to hosts, network elements and proxies. Proxies allow the existence of NSIS-unaware hosts, by carrying out signalling on their behalf, as illustrated in Figure 17

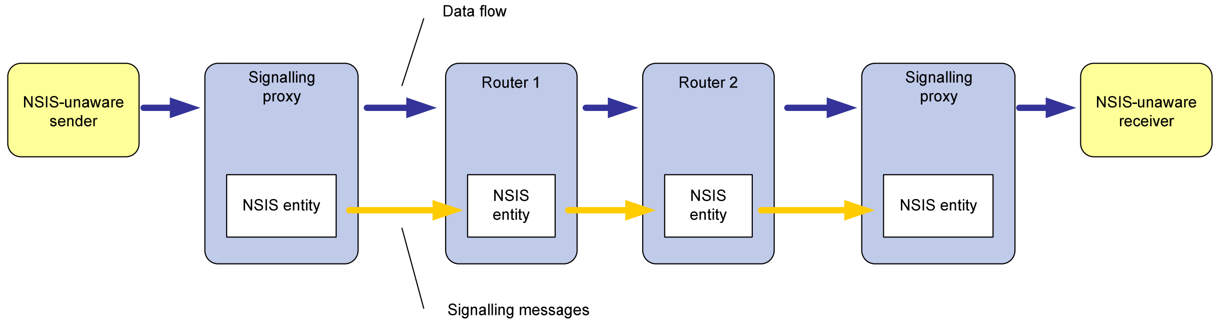

Figure 17: Signalling proxies

NSIS supports both on-path and off-path signalling. In the case of path-coupled signalling, signalling messages are routed through NSIS entities on the data path only, although between adjacent NEs, the route taken by signalling and data might diverge. In the case of path-decoupled signalling, messages are routed to NEs which are not assumed to be on the data path, but which are aware of it. In this case, the signalling endpoints may have no relation at all with the ultimate data sender or receiver.

GIST was designed as a soft-state protocol to manage all the messages and associations. Each time a state is entered or updated, a timer is setup or restarted. GIST has two main state tables: Message Routing State (MRS) and Message Association State (MAS). The MRS is responsible for managing individual flows and the MAS is responsible for managing associations between individual peers. When a timer expires (if no message is received for the corresponding flow or association) the state is automatically removed from the state tables. If a state is required again, a new handshake is needed and a new association must be created.

4.2.4 Misbehaviour and Anomaly Detection

The adoption of monitoring and security mechanisms in networks is currently the focus of substantial research. Network Intrusion Detection Systems (NIDS) are systems implementing mechanisms that are able to analyze traffic and obtain information that will help characterize the malicious activities. Usually these systems are based on information obtained from logs generated by packet analyzers. The detection method of such non-allowed activities uses attack signatures – i.e., patterns. These signatures are registries that represent exactly what we are looking for. For example, a signature may be based on the analysis of strings in the data field of captured packets. Whenever such string is detected in a packet, it will indicate a possible malware activity.

Even if the use of signatures in NIDS is the common place, recent works have showed that behaviour-based approaches constitute the next-generation tools. Such systems learn the normal behaviour of traffic and systems, and then continually examine them for potentially harmful anomalies and for a behaviour that frequently accompanies incidents. Usually such systems detect anomalies by looking at IP features (IP addresses, ports, flows), and upon identifying potential problems, the systems further analyze the traffic to determine whether an attack is indeed occurring or not. The latter can be carried out by comparing anomalies with entries in a dictionary of harmful behaviours.

These are the fundaments of NADA (Network Anomaly Detection Approach), a methodology being developed in order to be able to detect anomalies in traffic traces. To accurately accomplish its task, NADA uses a collection of anomaly signatures. However, the signatures being used are behaviour-oriented. This is possible by using information about traffic flows and its IP features to detect the anomalies, and not intrinsic characteristics of the anomalies. NADA objectives include:

1. Detection of anomalies; i.e., determining whether an anomaly is occurring.
2. Classification of anomalies; i.e., determining what kind of anomaly arises. This amounts to determining whether the anomaly is legitimate or not, and its precise kind (HTTP flash crowd, scanning, SYN flooding, etc.).
3. Identification of anomalies; i.e., being able to determine all the anomaly constituting packets and flows.

In addition, NADA aims at being completely generic, and able to work on any kind of time series associated with incoming traffic or packet traces. Currently NADA works over three different data time series: Number of packets per unit of time; Number of bytes per unit of time; and Number of new flows per unit of time. However it is extensible to any other kind of time series.

It is worth pointing to the need for working on several time series to ensure correct detection, classification and identification of anomalies, since each type of anomaly acts differently over each of the parameters presented above. In addition, as the NADA tool is expected to be adopted by network operators, it has to work on network and traffic representative features, such as bytes, packets and flows, and simple statistics. Times series about the number of SYN or RST packets could also be easily added to the algorithm, if they are meaningful for the operators. In any case, by using only simple mathematics, the goal is to make NADA easily and efficiently exploitable and configurable by network technicians.

4.2.5 Mobility and Nomadicity

Community Networking includes a broad set of technologies, most of them wireless, that when combined in a seamless way may offer a very rich networking environment supporting multimedia communications and, in general, content distribution networks. In this environment, we can identify the following technologies: WLAN (IEEE 802.11x), GPRS (EDGE), 3G (UMTS, HSDPA), WiMAX (IEEE 802.16), MBWA (IEEE 802.20), to name the most known current technologies. In the near future we may have new suitable technologies as well. The trend nowadays is to have several interfaces available in a single device, even in a small mobile device such as a phone or PDA. MIMO technology is emerging with a strong push. So, multihoming is a "natural" feature for mobile devices. The selection of the interface or interfaces being used at a given time is a challenging open issue.

From the point of view of network connectivity and management several technologies may be considered as, for instance, wireless LAN (WLAN), ad-hoc networks (MANET), mobile routers (NEMO), wireless mesh networks (WMN), wireless sensor networks (WSN or WMSN for multimedia sensors), and even satellite networks (mainly for content distribution in wide areas). Taking into account that nodes (fixed or mobile) may appear and disappear, be self-powered or not, be moving or not, act as a router or not, be a small multimedia sensor or a powerful multimedia device, it is evident that some of the characteristics of MANET, NEMO, WMN and WMSN may be mixed.

To complete this framework scenario, we may consider all the issues related to "handover" from one coverage area to the neighbour one. The community networking scenario adds extra complexity to the handover process within the same technology (horizontal handover), because now we must include the handover between two different networking technologies (vertical handover). As different wireless technologies are being deployed, word-wide heterogeneous networking is becoming a reality. This diversity of wireless technologies must be taken into consideration and it presents different research challenges.

In order to efficiently manage this heterogeneous environment the IEEE is developing the IEEE 802.21 standard. This standard aims at enabling handover and interoperability between heterogeneous network types, including both 802 and non-802 networks. The 802.21 standard defines an abstraction layer, providing Media Independent Handover (MIH) functions with the goal of simplifying the management of handovers to and from different access technologies. In this heterogeneous environment, mobile protocols such as Mobile IP, NEMO and MANET may benefit from the IEEE 802.21 standard.

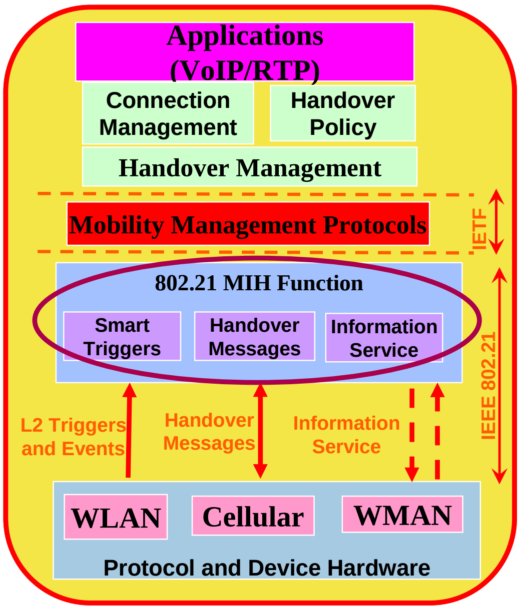

Figure 18: IEEE 802.21 Architecture

The following research challenges may be addressed:

- Seamless co-existence of all the wireless technologies is required, including seamless mobility. IEEE 802.21 plays an important role here.
- Cross-layering is a requirement for optimizing mobility and multihoming. This includes cross-layer issues between transport, network, MAC, and even physical layers. Some bridging functions at MAC level are required for seamless integration of networks and routing/MAC cross-layer design may help efficient and reliable routing.

- Management of Multi-hop communications should also be addressed.
- Self-configuration connectivity must cope with the changing availability of the nodes and compute new paths to the sink nodes (gateways to the Internet).
- Bandwidth balancing in multihomed devices amounts to deciding on which interfaces and paths to be used. The estimation of the available bandwidth and other quality of service parameters may be used to make this decision.

4.2.6 Overlays for Content Delivery

The traditional P2P Content Distribution infrastructures have shown a clear lack of mechanisms to ensure efficient and fair utilization of network resources. This well known weakness can easily evolve in main transport networks congestion, wasting key communication resources with critical impact on network performance, end to end quality of services offered as well as from an economical perspective.

This critical resources management issue has two main actors: the *Network Providers* that offer connectivity to the end users, and the *P2P applications* that make use of that connectivity to offer content. From the ***Networks Provides' perspective***, current networks were not envisioned or engineered for the traffic models imposed by P2P applications. P2P applications do not include mechanisms to minimize the cross-network traffic, increasing exponentially providers costs; besides that, these applications are neither able to deal with networks congestion, due to the abuse of UDP, or a large pool of TCP connections. Finally, providers have to deal with the fact that a small portion of customers are using a huge percentage of their network capacity, usually with a full-time use of their connections. The main consequence and risk to assess is an exponential increase of potential and severe network congestion, as most of those networks have been designed assuming statistical multiplexing.

From the ***P2P applications perspective***, P2P applications do not implement any algorithms to select peers with the best connectivity (i.e. considering delays, throughput, etc), because it is something difficult to figure out without information of the external networks topology. The main direct consequence is to get worst and suboptimal transfer rates; from end user perspective, this means quality of experience and satisfaction continuous decrease. In the last years, a few P2P applications have included mechanisms to try to detect and prioritize network traffic. They try to retrieve geographical and network proximity information, based on external services like Content Distribution Networks [14], or on round-trip-time delays measured explicitly to place peers in a virtual euclidean space [13].

As a solution, the *congestion* can be drastically reduced by sending additional feedback from the network to allow applications to reduce the traffic under congestion [12], or by marking P2P traffic explicitly to allow the network to discard those packets in a critical congestion scenario, giving them a lower than best-effort quality (i.e. via Diffserv). In the *peers selection optimization*, proposed solutions involve a server in the network (Figure 19), which provides topology information assisting the applications to select the best possible peer among the ones obtained from the tracker, or the distributed overlay. Other techniques to improve the performance of P2P content distribution, e.g. reusing the multicast capabilities of the network or introducing nodes caching contents in the operator network to reduce transit costs should be explored.

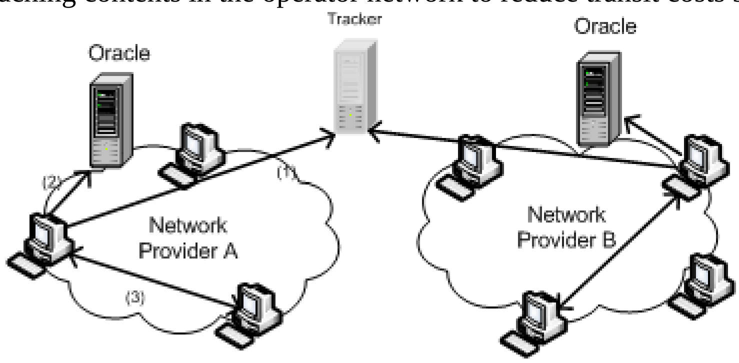

Figure 19 : Framework for Assisted Peer Selection

4.2.6.1 Efficient Peer Selection in P2P Live Video Streaming

It is widely accepted that in order to deal with loss of content in a P2P streaming system, peer selection should not only be done by considering network quality criteria but also the buffer status of the candidate parent peer. A whole series of new protocols base peer selection on whether or not the selected parent peer has the required data chunks. In order for peers to know about the buffer status of the other peers in the system "buffer maps" are frequently exchanged imposing a high control overhead. It is clear that it would be beneficial for a peer to be able to find required data chunks to a large number of other peers. This way the peer would have an increased probability to find peers with good connectivity between them while it would be able to initiate more than one connection executing parallel downloads of data chunks and by doing so increasing the reliability. This increased availability would be possible only if buffered content of different peers are highly correlated. Factors that affect this correlation should be investigated since as it can severely affect the performance of a P2P streaming system. Under limited buffer resources one of the major factors that affect correlation between buffered content among the peers in the system is the selection of the playout policy to be adopted.

Playout policies for video receivers have been studied extensively in the past for the client-server case, involving a single server and multiple, independently operating, receivers. P2P streaming systems, however, are fundamentally different. Besides rendering the received stream for the benefit of the local user, a receiver also acts as a sender and forwards it to other "downstream" receivers which, in turn, can forward it further down in a *hierarchy of peers*. In a client-server system each peer receives the stream directly from the sender and does not relay it any further. In P2P systems the new conditions have to be carefully factored-in when selecting playout policies for such systems, either when the same policy is adopted by each peer or when each peer autonomously selects its preferred policy. Overlooking them can easily lead to a totally inacceptable performance and even lead to the collapse of the system.

The *desynchronization of playout points* can have dire consequences on the probability of finding a better up-stream relay node when the current one is experiencing congestion or when it departs from the system without prior notice. Switching to a new relay node should not only provide a good connection between the client and the serving peers but also enables the serving peer to provide the next frames required at the client peer. If the serving peer has earlier frames in its buffered content than the once required this will cause a *discontinuity* in playback to the client peer i.e. a freeze until the required frames are available, while if only later frames are available this will cause the permanent *loss* of the required frames. The adopted playout policy affects the *correlation between the buffer contents of various peers* in the system and thus affects the *availability* of switching to a better up-stream relay able to provide a seamless hand-off without discontinuity in playback or loss of content.

A major conclusion of a number of studies is that globally synchronized playout in a P2P streaming system results in "positive correlation" of buffer contents among peers. This way peers may perform a seamless switch in times of poor reception in order to sustain continuity in playback. The adoption of such an approach in systems which are based on the exchange of "buffer maps" would greatly improve performance while it would eliminate the control overhead since availability will be high thus even a random selection of a list of parents would be sufficient. Furthermore the inherent advantage offered by the P2P architecture is preserved and exploited on the benefit of playout quality. The high performance, strict interactivity and low implementation cost make viable the operation of an IPTV service.

4.2.6.2 Distributed Selfish Replication under Node Churn

While network nodes can decide on their own on the content they will store to serve best their clientele, to utilize the P2P availability of storage, one approach is to form groups of nodes that are in close distance and can cooperate in order to collectively decide which content to store and then provide it to local or remote users effectively and economically. That is, a user's request is first received by the local node. If the requested object is stored locally, it is returned to the requesting user immediately, thereby incurring a minimal access cost. Otherwise, the requested object is searched for, and fetched from, other nodes of the group, at a potentially slightly (due to the proximity) higher access cost. If the object cannot be found anywhere in the group, it is retrieved from an origin server, which is assumed to be outside the group, thus incurring maximal access cost.

A key issue in content distribution is the object placement problem. Several object placement problems can be defined regarding a distributed replication group, which refers to the selection of objects for the nodes, under given node locations and capacities. They focused on the optimization of the so-called *social utility* (sum of the individual *local utilities* of the nodes, defined as the delay and bandwidth gains from employing replication), which is in some cases harmful to several local utilities. Another cooperation scheme manage individual and collective performance when better nodes cooperate in deciding which objects to store so that the cost of providing content to their clientele can be decreased, compared to that induced when operating in isolation. First nodes compute their greedy local object placements (i.e. replicating the most popular objects according to their local demand) operating in isolation. Then, each node takes a turn and improves its placement according to the placement of other nodes that have already improved their placements. The main result developing such a scheme has been that no node is mistreated (i.e., the average access cost for objects requested by its clients is higher than the cost incurred if it were not part of the group), and thus the scheme is sustainable. However, a key assumption has been that nodes are always available and that there is no *churn* - change in the set of participating nodes due to joins and leaves.

Since in almost every distributed system node churn is prevalent, a key issue is to find a scheme that can assure the mistreatment-free property. A strategy can be implemented in such a way that takes into account the probability of the nodes to be available in order not to introduce mistreatment problems. Clearly, such a consideration would require additionally that each node knows its probability and the probabilities of the other remote nodes to be available, before nodes can cooperate to replicate content. Such knowledge may be built in the system progressively or could become available by a trust and reputation mechanism. The challenge would then to devise mechanisms that ensure the mistreatment-free property in a distributed selfish replication group in the presence of node churn.

4.2.6.3 Content-based search for media in distributed environments

Today’s the Web pages are full of (2D and 3D) multimedia content. Podcasts are becoming a popular source of information. This trend is caused by a fact, that the users are not only able to consume (“consumers”) the content, but also are able to become producers (“prosumers”) themselves. The amount of media available in the traditional Web is growing at an enormous pace. This will be supported, in a very near future, by services based on the Peer-to-Peer paradigm.

Unfortunately, this growth of popularity of media is not accompanied by the rapid development of media search technologies. The most popular media services in the Web are typically limited to textual search, which is not efficient as it requires searching at metadata. A solution to the problem is to provide more natural querying methods – such as **Query by Example** (QbE) and **Query by Sketch** (QbS). Such methods will support the traditional ones. The new emerging search techniques should be not only applicable in the traditional, centralized services, but their development should foresee the imminent introduction of decentralized media access and delivery solutions based on P2P paradigm. The newly developed search algorithms should also support the emerging media formats and types – such as, for example, user generated 3D content, podcasts or P2P TV.

Distribution and enhancement of the media search services will allow for easier access for the citizens to the multimedia content. This will give a further boost to the community creativity. Distribution in a P2P manner allows also for reuse of the computing power of the individual PC’s. This allows deployment of the services, which would require enormous computing power if designed in a centralized way. Distribution of search and retrieval allows the user for better protection of their privacy and rights – they are no longer required to upload their works to the centralized services, such as e.g. YouTube, in order to disseminate them.

5 CROSS LAYER ADAPTATION FOR ENRICHED PQOS

One of the main challenges in today's networked audio/visual communication is the ability to provide a sustainable end-to-end quality as indicated by the user, throughout the entire duration of the service delivery. Offering QoS-based services involves interactions, not only among a number of entities along the service delivery chain, but also across different layers. To coordinate effective adaptation and mapping of QoS parameters at service, application and network layers, cross-layer interactions are required. The objective of this adaptation and interaction is to find a satisfactory QoS trade-off, so that each end-user's service can be supported with available network resources. However, a number of questions arise, including: "How to measure and evaluate the QoS?", "What feedback to give?", "What Adaptation to do?", "Which Entity will perform the adaptation?", "Where to do the Adaptation?"

In this chapter, we highlight a very important issue in streaming multimedia over the Future Internet: the cross layer adaptation issues in order to achieve an enriched PQoS.

5.1 Measuring & Evaluating the QoS

In order to measure and evaluate the QoS, a number of objective and subjective methods have been proposed. A subjective evaluation of the quality of the content is difficult due to various factors including time, cost and human perception. Strictly speaking, subjective tests require a large number of tests operated under controlled psychometric experimental conditions, to obtain statistically meaningful Mean Opinion Scores (MOS), summarizing the Perceived QoS (PQoS). Obviously this is not a good or even possible solution for real-time audiovisual services. As alternatives, objective measurements are used by analyzing the signals in both compressed (e.g. MPEG-4/H.264 compressed video stream) and non-compressed (e.g. reconstructed RGB video as the output) formats.

As illustrated in Figure 20, from the end users' perceptual experience to adaptation decision, a series of QoS are involved: Subjective QoS, Objective QoS, Network QoS (**NQoS**), and Adaptation QoS (**AQoS**). However, more types of QoS (and PQoS) may be defined in an end-to-end environment.

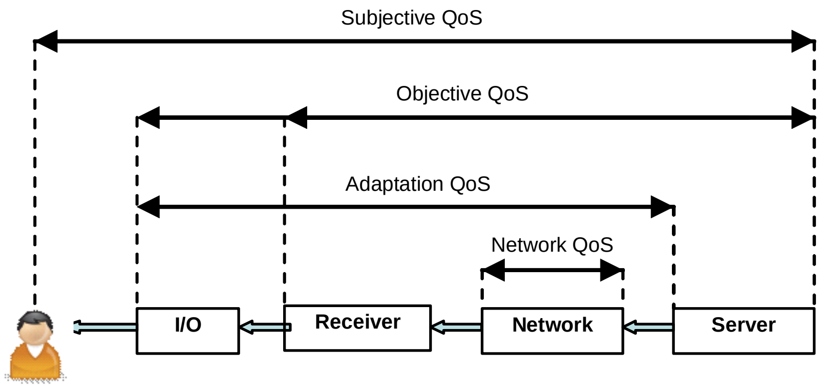

Figure 20: Subjective, Objective, Application, Network QoS

At the service layer, ***Expected PQoS*** is defined as the perceived quality that an end-user expects from a contracted service, described in a customer Service Level Agreement (SLA). This may be determined by the customer's previous experience of similar services or by the service provider's offering. The *Expected PQoS* depends on the user's preferences and the capabilities/constraints of the utilised terminal.

Different value-added applications have different performance needs and constraints. *Application QoS* (***ApQoS)*** is defined as the application quality to express these needs and constraints in technical terms. *ApQoS* is characterized by encoding and transmission parameters, such as frame rate, resolution, coding format, latency, latency variation, loss ratio, etc. Based on the ApQoS bound and user expectation (*User Expected ApQoS*), the *Adopted ApQoS* is chosen to classify the application in terms of its quality requirements. If there is no application class available to support this user request, the user QoS preference must be changed and ApQoS bound may need to be adjusted.

The *Delivered PQoS* is the quality that a customer or end-user actually perceives when consumes the service at his/her terminal. The *Delivered PQoS* related parameters could be acquired during service delivery by subjective/objective measurements. The result is used for service monitoring and service fulfilment purposes including QoS adaptation. The *Delivered PQoS* monitoring is normally performed at level of an individual application stream. For scalability reasons, the NQoS (i.e., *Measured NQoS*) may be measured for a specific QoS class at aggregated level.

It should be noted that care should be taken in deducing *Delivered PQoS* from *Measured NQoS*, as different mapping models may need to be employed since users utilise various terminals in which user's *Expected PQoS* is different from one terminal to another.

5.2 Single-Layer Optimization/Adaptation

Today's Internet architecture is based on the typical TCP/IP and UDP/IP models. Layered architecture served as a backbone for the proliferation of networks due to its clean modular structure in which layers are implemented in complete isolation. But optimization of these protocols and mechanisms becomes the primary concern due to the inception of heterogeneous networks, terminals and real time delay sensitive applications. In response, optimizations strategies are proposed to improve QoS, increase throughput, and efficient utilization of bandwidth.

In a layered architecture, each layer has a set of distinct mechanisms and associated parameters to fulfil its functionality. Table 1 classifies useful parameters and adaptation mechanisms by their respective layers. Parameters presented in this table include both tunable and read-only parameters.

Table 1 : Mechanisms and parameters at different layers

Layers	***Mechanisms (to optimize)***	***Parameters***
User	User priority selection	Terminal characteristics, objective quality metrics (e.g. PSNR, VQM, SSIM), Subjective quality metrics (e.g. MOS, SAMVIQ, DSIS)
Application	Transrating, Transcoding, Forward Error Correction (FEC), Automatic Repeat Request (ARQ), Adaptive encoding/decoding	Rate, Codec, Protection level
Transport	TCP Congestion Control, UDP, Header Compression	Packet loss information, receiver window, congestion window, retransmission timer
Network	Packetization, DiffServ, TE	IP packet size, DiffServ Code Point, Handoff information
Data Link	MAC Protocols, Radio resource control, FEC, ARQ, Framing	Retransmission attempts, Error rate, retry limit, RTS/CTS, Handoff, Traffic classes, TDMA time slots, OFDM carriers
Physical	Channel modulation and coding	BER, signal strength, transmission power, capability profile
Context Information	Dynamic Voltage Scaling, Scheduling	Battery status, Architectural capability profile

5.3 Cross-Layer Optimization/Adaptation

During the last couple of years, it has been shown that adaptation techniques limited to adaptation within a single layer are deficient in providing global optimal setting for the system. In contrast, cross-layer approach has been extensively discussed in recent research literature for its viability for providing better performance than traditional layered architecture [17][18]. Cross-layer approach increases interaction among different layers to exploit the inherent characteristics of underlying network to maximize the utility (e.g. QoS) and reduce the cost (e.g., battery life, bandwidth). The involvement of multiple layers in cross-layer adaptation is important otherwise various mechanisms available at different layer are likely to counteract each other's effect.

Although cross-layer design emerged as a by-product of recent proliferation of wireless networks having totally different properties from wired networks, it offers various opportunities for heterogeneous environment, where a variety of application types, network technologies and terminal capabilities are utilised. Initial motivation to work on cross-layer problem was primarily derived from following reasons:

- Wireless networks are characterized by high bit error rate due to fast fading, co-channel interference and shadowing. To overcome these issues, different layers can cooperate to make transmission more resilient to noise. Error resilient A/V codec, error concealment and protection techniques at link layer and application layer, measurement of BER at physical layers and other mechanisms should help.
- Effective network condition estimation requires parameters from multiple layers e.g. Packet loss ratio, BER, SNR etc. Network condition estimation is necessary to increase utilization and reduce cost
- Low efficiency of transport protocols over wireless networks due to their inherent characteristics is also a reason for the consideration of cross-layer design.
- Heterogeneity of applications, terminals and networks require more rigorous adaptation mechanisms. Especially, in the context of multimedia services, content adaptation is absolutely necessary due to enormous dependencies arising from heterogeneity. Cross-layer adaptation can play a key role in handling such multiplicity of dependencies.
- Cross-layer adaptation can assist smooth transition of Internet from best effort to QoS.
- Apart from their unique characteristics, wireless networks support new modalities of operation that could not be supported by traditional layered architecture like multi-receptions capability.

5.4 MPEG-21 Multimedia Framework

As many actors are involved in delivering multimedia content from the provider to the consumer, interoperability plays a crucial role and calls for a multimedia framework. A comprehensive framework that deals with all these issues is MPEG-21 [16]. The aim of MPEG-21, the so-called Multimedia Framework, was to standardize a framework for ubiquitous multimedia consumption across heterogeneous applications and domains, e.g., networks and devices. All parts of MPEG-21 address a distinctive set of requirements, which allow implementers of the standard to design and implement a system or application that goes beyond simple multimedia content delivery in an interoperable way. MPEG-21 offers comprehensive description formats for the entire delivery chain:

- Content and service providers are able to describe their contents/services in an interoperable way if used with other content description frameworks like, for example, MPEG-7 [31] or TVAnytime. Furthermore, MPEG-21 offers means for describing and capturing the user profile and preferences.
- Network providers and device manufactures are able to describe their conditions and capabilities (i.e., context) which can be used by service providers for configuring/personalizing their services also. These description formats can be used for managing this heterogeneous environment.
- Finally, MPEG-21 provides support for Digital Rights Management (DRM) which can be used for authentication, authorization, and accounting (AAA).

The MPEG-21 standard provides the transaction of Digital Items among Users. A ***Digital Item*** is a structured digital object with a standard representation and metadata. As such, it is the fundamental unit of transaction and distribution within the MPEG-21 multimedia framework. In order words, it aggregates multimedia resources together with metadata, licenses, identifiers, Intellectual Property Management And Protection (IPMP) information, and methods within a standardized structure. A ***User*** is defined as any entity that interacts within this framework or makes use of Digital Items. It is important to note that Users may include individuals as well as communities, organizations, or governments, and that Users are not even restricted to humans, i.e., they may also include intelligent software modules such as agents.

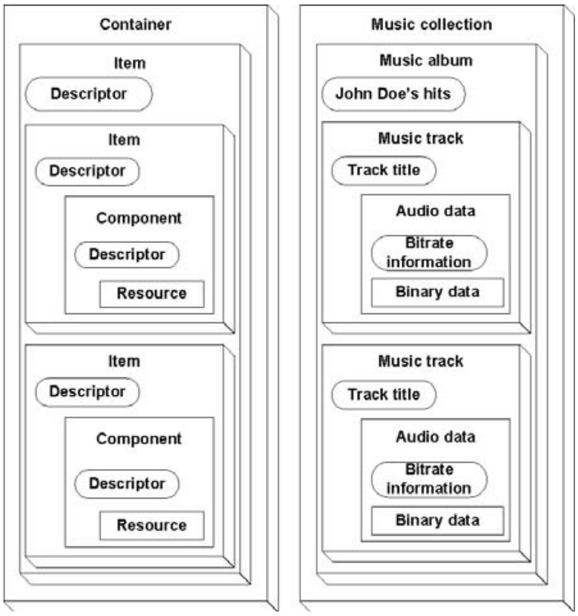

Figure 21: MPEG-21 Digital Item Declaration

A Digital Item can be thought as a virtual structured digital container of media resources and metadata. It can include resources of any media type: audio, video, text, images, and so on. Metadata is the related information for the entire DI or part of a DI which provides semantic support, as shown in Figure 21. The introduction of Digital Item changes the way of content handling and usage, from a single media resource to a composite media package, based on which standardized protection, adaptation, etc. are operated. So as to support the definition of the fundamental unit DI, MPEG-21 specifies Digital Item Declaration (DID) to provide a uniform and flexible abstraction and inter-operable schema to represent DI, and Digital Item Identification (DII) to ensure a unique and persistent identification of the DI and its resources.

5.4.1 MPEG-21 Digital Item Adaptation

Besides the inter-operability, digital content needs to be adapted to various transmission channels and terminal devices for delivery. Digital Item Adaptation (DIA) can be achieved by applying various approaches such as adaptation at the server side, at the intermediate proxy or at the terminal. We list here all the relevant requirements exposed to the terminal side from these adaptations:

1) **Device independence adaptation:** From a terminal's perspective, terminal-independence adaptation is usually employed. *User Environment Description (UED)* is the key of this approach. It includes descriptive information related to user characteristics, (e.g., user information and user preferences), terminal capabilities (e.g., codec capabilities and display capabilities), network characteristics (e.g., available bandwidth, delay, and error), and natural environment characteristics (e.g., location and time).

2) **Content dependence adaptation:** such approach relies on the coding scheme which provides scalability. Particularly, in the case of SVC, it has achieved temporal, spatial and quality scalabilities co-existing in a single bit stream. This allows video adaptation at bit stream level. Such benefit outperforms other coding schemes as it increases the adaptation flexibility. For example, if a terminal is limited by certain constraints, e.g., computing memory or power, and its decoder can support SVC, there is no need of

intermediate adaptation, since the receiver can perform the adaptation itself by discarding the relevant Network Abstraction Layer (NAL) Units that convey enhancing layers.

3) **Adaptation by quality constraints:** to achieve optimal parameter settings under certain constraints imposed by terminals and/or networks for QoS management, Adaptation QoS (AQoS) is provided to assist the adaptation engine for decisions. AQoS specifies the relationship among various constraints, feasible adaptation operations satisfying these constraints, and associated qualities. AQoS can be used together with *User Constraints Description (UCD)* to acknowledge the adaptation engine (Figure 22).

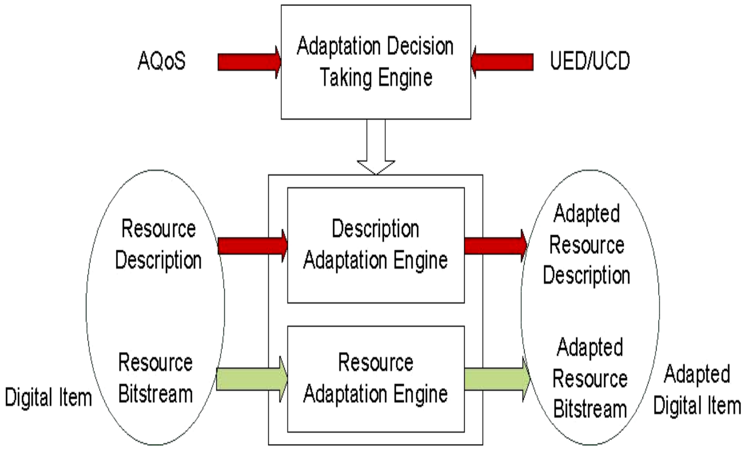

Figure 22: Digital Item Adaptation

The above adaptation put forward several requirements (and possible approaches for solutions at the same time) for a terminal: 1) UED/UCD functional modules needs to be integrated in terminals; 2) a media decoder with SVC codec support; 3) terminal and network QoS management for AQoS need to be provided.

5.4.2 MPEG-21-enabled Cross-Layer Adaptation

The idea behind the concept of MPEG-21-enabled cross-layer adaptation is to perform coordinated actions across several layers along the end-to-end content delivery chain while maintaining interoperability. The concept of MPEG-21-enabled cross-layer adaptation can be described as below [19], [20]:

1. ***Cross-Layer Model (XLM)*****:** The XLM provides means for describing the relationship between QoS metrics at different levels – i.e., PQoS, ApQoS, and NQoS – and layers – i.e., according to the well-known ISO/OSI reference model – which may cooperate to improve the ability of applications to ensure certain objectives such as QoS guarantees, power saving, users preferences, etc.

2. ***Instantiation of the XLM*** by utilizing ***MPEG-21 metadata***: Description formats (i.e., tools) as specified within MPEG-21 Digital Item Adaptation are used to instantiate the XLM for a specific use case scenario, e.g., Video-on-Demand. In particular, the *Adaptation QoS* (*AQoS*) description tool is used as the main component to describe the relationship between constraints, feasible adaptation operations satisfying these constraints, and associated utilities (qualities). The *UED* tools are used to describe the context information where Digital Items are consumed in terms of network conditions, terminal capabilities, user preferences, and conversion capabilities. Finally, the *UCD* tools are used to express limitation and optimization constraints.

3. ***Cross-Layer Adaptation Decision-Taking Engine (XL-ADTE)***: The XL-ADTE is the actual subsystem which provides the optimal parameter settings for media resource engines according to the XLM by processing the metadata compliant to MPEG-21 DIA. In other words, the XL-ADTE is a generic (software) module that solves optimization problems expressed by using MPEG-21 DIA-based metadata according to the XLM.

Within the end-to-end multimedia *delivery chain*, the network QoS may be measured on an aggregated level and mapped to PQoS of individual streams [21][22]. The monitoring system may utilize an end-to-end QoS monitoring system [23] following the of IETF's NSIS architecture [24].

5.4.3 MPEG-21 Open Issues

It has turned out that only relatively small portions of the whole MPEG-21 framework have been adopted by industry so far. This ultimately leads to the question whether MPEG has addressed the requirements in a vital way and what needs to be done to foster adoption of the MPEG-21 concepts on a broader scale:

- An important issue is the link to the transport/delivery layer. That is, bindings to the transport layer and an interoperable description format of the capabilities of the delivery layer itself. However, this is usually out of scope of MPEG and calls for actions to be taken within other standardization bodies.
- “The future of digital media computing is meta” [32] but it seems it is not as industry adoption of standardized metadata formats is relatively slow compared to the adoption of coding formats (e.g., MPEG-1/-2/-4). Proprietary formats are still predominant or the industry does not see an added value for using interoperable formats which calls for the need of new “metadata-centric” business models.

5.5 Challenges in Cross-layer Adaptation

The concept of cross-layer design sounds persuasively appealing. However, the successful experience of layered architecture burdens the adoption of a cross-layer approach. Currently, the research community is endeavoring following challenges in this paradigm.

1. Cross-layer adaptation of the complete network infrastructure is very intricate due to handling enormous dependencies possibly in real time. A flexible architecture with proper interfacing between the layers is inevitable.
2. Cross-layer design breaks the layers and hence a clean isolated implementation of different protocols is no longer possible. Each cross-layer approach affects a complete system. An analysis of these effects becomes difficult.
3. The effects of coexistence of different cross-layer interactions are to be observed on system performance, maintenance and scalability. Analysis is further perplexed if different type of cross-layer optimizations are deployed across an end-to-end delivery chain. Compatibility of different cross-layer adaptation is one of the major issues. Due to lack of research in this dimension, cooperative behaviour of multiple cross-layer approaches is highly indeterminist.
4. Global metrics are required that maximize the utility (e.g., QoS) and minimize the cost (e.g., battery life) under various constraints by efficiently prioritizing layers' local optimization criteria.
5. Optimization of cross-layer parameters is a complex multivariate problem with various constraints derived from QoS guarantees, available bandwidth, power consumption, etc. Solutions of such optimization problems converge in multiple iterations. Apart from the essential requirement of computational efficiency, the highly dynamic nature of wireless networks demands a rapid convergence of the solutions. Moreover, objective functions for maximum users' satisfaction have to be further investigated.
6. It has to be evaluated where the actual control of a cross-layer adaptation should be located. Without a central control, different cross-layer adaptations might counteract each other. Different candidates include a separate coordinator or a particular OSI layer.
7. Cross-layer adaptation simulations are generally more complex than traditional network simulations. Hybrid approaches combining network simulation tools, hardware support and analytical approaches are usually required
8. Not even a single cross-layer proposal has been tested comprehensively under real world traffic scenarios and hence QoS, power consumption and scalability of these approaches are yet to be gauged deterministically.
9. The assurance of fairness is yet an un-promised reality by cross-layer design.

6 MULTIMEDIA RIGHTS MANAGEMENT

Multimedia delivery platforms of the future internet are intimately intertwined with Multimedia Rights Management, aka Digital Rights Management (DRM) or Content Protection (CP). Indeed, while certain multimedia content is sure to be provided for free or semi-free (e.g., financed through personalized advertising), effective management of multimedia rights and licensing the rights to processing and consumption of multimedia properties are vital to a wide range of business models thriving and enabling the future internet. In fact, the underlying premise of any ad-supported semi-free business model depends on conditional rights licensing and on built-in means for content usage tracking and consumption patterns.

This section describes the multimedia delivery platform rights management and content protection framework for the future end-to-end multimedia delivery platforms, and reviews several relevant international standard bodies and industry forums, which address the issue of formalizing a standard approach and specifications for DRM.

The multimedia delivery platform rights management architecture needs to be a versatile general-purpose architecture which covers a wide variety of content distribution services and content source types. In particular, it needs to address the enabling of end-to-end QoS delivery platforms and the complexity of requirements imposed by assured e2e QoS affecting rights management.

Additionally, one of the desired objectives is to provide a standards-based, loosely-interoperable DRM framework for end-to-end content protection and digital rights management with QoS management. Several standards (international and industry) have made significant progress in recent years. We study the relevance of three of the leading DRM standards – MPEG-21 IPMP, OMA and DMP (the Digital Media Project). This section also provides our commentary on the prospects of turning the intricate DRM interoperability approach offered by DMP, into a widely-accepted framework.

6.1 Comparative Analysis

Currently the major DRM schemes include: MPEG-21 IPMP, MPEG-21 REL, OMA and DMP. We may position those standards, in comparison to one another and in respect to their creation objectives, as follows:

a) Several of those standards put DRM Interoperability at the base of their set objectives; however DMP is the only industry body who set out on the ambitious task of achieving true system-wide interoperability among ‘foreign’ DRM systems.

b) DMP deals with the complete framework. It provides a full ontology, describing not only the tools but also the entities and how they interact, however DMP is inclusive in the sense that it does not limit the adopter to which specific languages (=standards) to use. DMP accomplishes its DRM interoperability objective by defining tools and use cases covering the complete value chain in a comprehensive manner and inclusive manner, which deals with all aspects of the rights management and handling of content described in its eight distinct models: Creation Model, Distribution Model, Delivery Model, DRM Tool Model, Domain Model, Device Model, Import/Export Model and Data Model. While not a DRM standard itself, it provides specification of tools and processes which may be adopted by DRM standards in order to achieve interoperability.

c) Each of the (other) standards analyzed provides only a portion of the framework. MPEG-21, with its IPMP, REL and DID standard definitions, has a clear advantage over alternative DRM standards, since it covers a wider portion of tools and (to borrow from the DMP terminology) models, as well as since DMP builds upon familiar MPEG-21 concepts. Yet, it is not comprehensive. MPEG-21 provides the key tools and metadata for DRM, however it does not concern itself with the overall functionality. Hence, it does not offer the same richness of tools as DMP, nor the same end-to-end backbone. For example, MPEG-21 does not describe the content owner entity, but rather the representation of the DRM descriptors and tools serving content creation (data description, tools, license, etc.). As such, it provides the *creation model* in a limited form, not inherently covering adaptation as an example.

d) MPEG-21 is inclusive in the sense that it can support multiple types of tools. ISMACryp is one instance of such class of MPEG-21 IPMP tools, limited in functionality to the encryption / decryption method. The signalling defined within ISMA is extremely limited, and – although interoperability appears as part of its objective, its benefit is limited to the lowest level, hence it covers part of the value-add chain, while most specific and exclusive in its capabilities (exemplified by its choice of algorithm, etc.).

e) OMA parallels MPEG-21 in its positioning. Not as elaborate as MPEG-21 REL in its licensing capabilities, OMA covers more or less the same scope of DRM *models* as MPEG-21.

6.2 Rights Management Challenges

One of the approaches used in the battle against digital media piracy is to use counter technology, known as Media Rights Management. Various rights management technologies exist, concentrating on user and content authentication and enforced license rules. The problem is that currently existing DRM technologies work like chemotherapy against cancer, which kills good cells along with the bad ones. Exercising proprietary techniques, limited in scope and inflexible, current DRM technologies diminish the potential of digital media. They cause prices to increase and accessibility to diminish. Moreover, by attempting to protect the delivery channels rather than the content, current rights management technologies often fail to achieve their target. A single copy stolen from a protected channel can be, within days, distributed for free to half of the Internet.

The limited success of copyright enforcement in the media market can be attributed to the ease of copyright violation, and the high motivation of this violation. Successful enforcement system must lower the violation motivation and block the technology loopholes.

Obviously the lower it costs to stay honest the less is the motivation not to be. But this is not only a question of price. A fundamental change of culture is required in order to create a global convention that royalty payment is normative and fair.

To cause copyright enforcement to look “normative and fair” two principal conditions have to be met:

- Rights management must favor the interests of creators and consumers rather than those of merchants, agents and lawyers.
- The simplicity and robustness of access to protected and non-protected content should be similar. The only difference between protected and non-protected content should be the price.

These two conditions may seem to conflict with the other requirement of rights management – security.

The future internet multimedia delivery platform is challenged to square that circle, while supporting its overriding functionality of securing end-to-end QoS. The goal is to develop a platform that will favor the interests of the multimedia content creators and the multimedia consumers and will be secure, yet simple and non-restrictive. To achieve that, the following principles are used:

- **Protect the item – not the channel.** The reliance on proprietary technology forced rights management to focus on protecting delivery channels that are wholly under control of the technology licensor. It is the channel that is protected, rather than the media. Once a media item is retrieved out of the channel, it is available in plain form and can be, and practically is, easily distributed over unprotected channels.
- **Open standards.** A truly efficient protection system must focus on the media. The access to a media item should be controlled, regardless of distribution path and consumption method. Since media items can change hands and be routed between equipment and systems of different manufacturers, the key to successful access control is interoperability. And the key to interoperability is open international standards.
- **Affordable technology.** Keeping low prices is one of the most crucial means in the effort to motivate users to pay for content. This can succeed only if the rights management technology itself does not

increase the cost of media access. This is another reason for using standard technologies. By law of nature proprietary technology is always expensive.

- **Impersonation.** In the old analogue days buying a licensed media item was simple – buying the physical media. This does not apply in the digital days when there are practically no physical media. Digital licenses are attached to other entities – devices or persons. However associating a license to either is problematic, as it contradicts one basic requirement for DRM – that users can have with digital licenses same rights as they had with analogue ones. Those rights depend on impersonation – attaching the license to a virtual identity.

  This indeed is one of the biggest challenges – to develop virtual identities that cannot replicate without a license, but can easily and simply move between real identities - people and devices. It must also include transfer of licenses between devices of different classes, e.g. set-top boxes and mobile phones.

- **Rights Management Interoperability.** MPEG-21 is the first standard that treats a digital media item as a tradable entity and gives the necessary tools for truly open and efficient exchange of items. With MPEG-21, a creator can produce an item, pack it (DID – Digital Item Declaration), “print” its description on the pack (DII – Digital Item Identification), add usage instructions (DIA – Digital Item Adaptation) and even specify terms of usage rights (with RDD/REL – Rights Data Dictionary/ Rights Expression Language). A digital signature can be used to authenticate the source of the item and ensure persistent association of the description with the item.

  With these capabilities, MPEG-21 has the potential of increasing the market reach of digital media and ease access to consumers, while lowering the delivery cost and guaranteeing a fair distribution of royalties.

Digital media and MPEG-21 have the potential of catalyzing a new, and fair, sense of rights management interoperability and globalization. Instead of the global resources being used to enrich the few, digital media can proliferate in a world where frontiers are removed, supply channel shortened, and fair global competition, based on creativity, is encouraged to the benefit of consumers and artists.

DMP (Digital Media Project) is an activity that can be viewed as complimentary to MPEG-21. While MPEG-21 provides tools to support interpretability, DMP advises how to use these tools in order to achieve interoperability.

The future multimedia delivery platform will make an intelligent use in rights management interoperability to achieve the grand vision of “anywhere, anytime, to everybody”.

7 LOOKING FORWARD

There is a rapidly growing momentum behind worldwide broadband deployment and the emerging convergence of voice, video and data services. It is important to note, however, that streaming multimedia content in the Future Media Internet is very different from traditional cable/satellite TV services and will place significant new demands on telecom network infrastructure. Users will have a much wider channel selection and will ask to watch whatever they want, whenever they want it.

The future Internet, with its anticipated continued exponential growth and emerging dynamic ambient usage patterns, poses a formidable challenge to the Media Delivery Platforms designed to effectively carry the hundreds-of-Exabytes of Networked Media content, in ways not all fully foreseen at the present time. User connectivity, new media sensations, gaming, user participation / user communities, search tools and internet advertising are among the key evolutionary and revolutionary business trends now foreseen.

The challenges facing the future media delivery platforms vary from capacity, bandwidth efficiency, scalability, quality of experience and responsiveness through seamless support for the emergent varying business models and usage interaction models all the way to internet-of-things, machine-to-machine communication (including RFID), mobility, personalisation, accessibility, age and gender inclusion, assisted living, information explosion, reliability, affordability, information integrity, revenue generation, trust and rights management.

One of the greatest obstacles in the successful implementation of Future Media Internet from the Media Delivery platform point of view is the converged network architecture and the transport network. As currently constructed, the converged wireless and wireline network architecture supports Internet traffic, VoIP and video over the Internet, but the perceived quality-of-experience (PQoE) for each of these services is limited. For IPTV to succeed, carriers and service providers must offer a PQoE equal to or better than that offered by today's cable/satellite TV. The key to achieving this goal will be the development of content-aware networks or at least content-aware edge devices, which are capable of tracking, managing and prioritizing the multiple signal streams flowing through the edge of the network.

Mission-critical Service Oriented Architectures, real-time Event-Distributed Architectures, intelligent content-aware services, content discovery and content aware routing are additional key issues of the Future Media Internet. Today, the dominant routing protocols in the Internet such as Open Shortest Path First (OSPF) and Border Gateway Protocol (BGP) are capable of routing packets based on their IP addresses. However, these protocols have no knowledge of which server is suitable for a particular content or how to best route the media content in order to achieve significant improvements in the PQoE. Foreseen content delivery networks will be able to route different types of content, among different routes and reserve resources without user or application level signalling. In this respect, new open architectures and technologies for converged and scalable, seamless streaming services are required.

A key challenge for the Future Media Internet is the wide adoption of 3D video streaming. Although the 3D media technology exists for a couple of years, only recently advances in capturing, processing, displaying and networking have turned it into a reality for the large majority of users. For years, one of the major barriers to the 3D media has been 3D scanning and modelling. Today, there are many techniques for creating 3D models, but depending on the geometry and the material characteristics of the object or scene, one technique may be much better suited than another. Once 3D data have been acquired, further processing is needed. For example, improvements in automatic decimation, solving large 3D puzzles automatically, exploiting shapes in combination with texture information, level-of-detail (LoD) processing. All these can also be expected to greatly benefit from a semantic understanding of the data. After 3D content preparation, open architectures and technologies will be required for searching, streaming, caching, filtering, aggregation and presentation of 3D content with optimised PQoE and in-network content adaptation.

Future Media Internet should also inherently offer content protection, aiming to target all identified types of networks, content formats, services and applications. Content protection will go far beyond the legacy DRM schemes and try to focus on adaptive consumption models, authentication of distributed media content objects, user privacy, security and controlled content access under varying conditions.

New formats for multimedia content will evolve and emerge. From today's H.264 AVC, AAC, MP3, and early instantiations of SVC, the media delivery platforms will accommodate the carriage of a wide range of the above formats as well SVC, MVC, a multitude of audio and gaming-friendly formats, H.265, MPEG/Laser and other surprising industry standards and ad-hoc media formats. All this while striving to be on one hand content-agnostic, yet applying a network intelligence, achieved through intimate content awareness, for the purposes of traffic shaping, PQoS/PQoE, security, reliability and more, a tough challenge. Furthermore, the prevalence of virtual and parallel personalized worlds, coupled with progressively changing virtual characters, adds a dimension of complexity tricky to contain and to scale up.

Finally, existing networks' cross-layer control (CLC) and adaptation provides significant improvements in the PQoE under specific networking and transmission conditions. However, further research is required especially in the case of P2P topologies, where the physical infrastructure may be an arbitrary, timely varying combination of links belonging to different networks. Moreover, CLC schemes are required to face the network and terminal heterogeneity and take advantage of new (3D) advanced coding and delivery schemes by proposing network abstraction mechanisms, able to model the underlined end-to-end paths, describe the functional dependencies and determine the optimum adaptation of the multimedia resources.

The Content Aware Media Delivery Platforms of the future, properly researched, conceived and designed today, will deliver on the promise to evolve the Future Internet into a powerful medium for transacting business, sharing and distributing multimedia rights-managed content and services with superior quality, reliability and flexibility, and in the process improving citizens' quality of life, working conditions, edutainment and safety.

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9 LIST OF ACRONYMS

AC	Access Controllers
ADSL	Asymmetric Digital Subscriber Line
ADTE	Adaptation Decision-Taking Engine
AQoS	Adaptation Quality of Service
AS	Autonomous Systems
AVC	Advanced Video Coding
BGP	Border Gateway Protocol
CABAC	Context-based Adaptive Binary Arithmetic Coding
CAPWAP	Control And Provisioning of Wireless Access Points
CP	Content Protection
CSP	Content Service Provider
DI	Digital Item
DIA	Digital Item Adaptation
DIMS	Dynamic interactive Multi-media Scene
DMP	Digital Media Project
DRM	Digital Rights Management
DVB-x	Digital Video Broadcasting-x (where x is S:Satellite, C:Cable, T:Terrestrial, H: Handheld)
E2E	End to End
EDCA	Enhanced Distributed Channel Access
EPC	Evolved Packet Core
FTV	Free viewpoint TV
FVV	Free viewpoint Video
GIST	General Internet Signalling Transport
HmD	Home DOmain
HSDPA	High-Speed Downlink Packet Access
HSPA	High-Speed Packet Access
HSUPA	High-Speed Uplink Packet Access
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IPMP	Intellectual Property Management and Protection
IPTV	TV over the IP protocol
ISP	Internet Service Provider
JSVM	Joint Scalable Video Model
JVT	Joint Video Team

LASeR	Lightweight Application Scene Representation
LS	License Server
LTE	Long-Term Evolution
MBMS	Multimedia Broadcast Multicast Setvice
MCTF	Motion Compensated Temporal Filtering
MDP	Media Delivery Platform
MGS	Medium Granularity Scalability
MIMO	Multiple Input Multiple Output
MOS	Mean Opinion Scores
MPEG	Motion Picture Experts Group
MVC	Multi View Video Coding
NADA	Network Anomaly Detection Approach
NAL	Network Abstraction Layer
NIDS	Network Intrusion Detection System
NLOS	Non-Line-Of-Sight
NQoS	Network Quality of Service
NSIS	Next Step In Signalling
NSLP	NSIS Signalling Layer Protocols
NSP	Network Service Provider
OE	Overlay Entities
OFDM	Orthogonal Frequency Division Multiplex
OFDMA	Orthogonal Frequency Division Multiplex Access
OMA	Open Mobile Alliance
P2P	Peer To Peer
PDA	Personal Digital Assistance
PEK	Program Encryption Key
PHY	Physical Layer
PMP	Point to MultiPoint
PQoE	Perceived Quality of Experience
pSLS	Provider Service Level Specification
QbE	Query by Example
QbS	Query by Sketch
Qo	Quality of Service
QoSR	Quality of Service Routing
RAN	Radio Access Networks
RAT	Radio Network Technologies

RDD	Rights Data Dictionary
REL	Rights Expression Language
RME	Rich-Media Environment
RS	Reed-Solomon
RSVP	Resource Reservation Protocol
SAE	Service Architecture Evolution
SAF	Simple Aggregation Format
SEK	Service Encryption Key
SLA	Service Level Agreement
SMK	Subscriber Management Key
SNR	Signal to Noise Ratio
SP	Service Protection
SRC	Smart Route Control
STB	Set-Top Box
SVC	Scalable Video Coding
TEK	Traffic Encryption Keys
UCD	Universal constraints Description
UED	Usage Environment Description
UEP	Unequal Error Protection
UMB	Ultra Mobile Broadband
VCEG	Video Coding Experts Group
VOD	Video on Demand
VoIP	Voice over IP
WiFi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access
WLAN	Wireless Pocal Area Networks
WMN	Wireless Mesh Networks
WTP	Wireless Termination Points
XL-ADTE	Cross-Layer Adaptation Description-Taking Enginee
XLM	Cross-Layer Model

10 LIST OF CONTRIBUTORS

Based on work that has taken place in a number of EC co-funded projects, in FP6 and FP7, a group of experts have voluntarily contributed in this white paper aiming to describe the status, the state-of-the art, the challenges and the way ahead in the area of Content Aware media delivery platforms.

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