001930 BROADWAN

Deliverable D11

Service validations and demonstrations on heterogeneous networks based on IPv6

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Abstract:

This report outlines the plans for demonstrating and testing the services and applications that were or will be introduced by the partners during the BROADWAN project.

The report focuses on how to successfully demonstrate different services provided on IPv6 based heterogeneous broadband wireless networks.

The primary goal is to prove the technical viability of the new technologies. This procedure is presented here. Financial viability will be investigated later in D26.

Keyword list:
Testing, broadband service, service platform, EoD, VoD, video surveillance, advertising, online game, peer-to-peer architecture, storage system, collaborative business environment, rural areas, satellite access, distance learning, multicast, IPv6.

Executive Summary

As the research on future user, services and system behaviour and development has shown in Deliverables D6 and D7, we can expect in the upcoming 5-10 years a major increase in the volume of network trafficking. Since BROADWAN aims at providing a feasible broadband networking architecture for all the citizens of Europe, it will necessarily have to demonstrate the capabilities of the newly developed techniques and models.

This deliverable deals with the definitions of demonstration and testing procedures. In other words this document defines *who* will demonstrate *what*, *where*, *how* and *when*.

In this report we present the preparations for the validation procedure. The goal of this procedure is to demonstrate broadband services at three different areas in Europe and perform advanced tests:

- The demonstration platform at Limoges and Paris
- The demonstration platform in Spain
- The demonstration platform in Oslo
- The test-beds in the UK and Austria

The services include video on demand, gaming on demand, e-learning, and peer-to-peer networks. These services have one thing in common: they manage complex and largely dependent data of large quantities. In order to manage these data (i.e. distribute to the users) efficiently, one needs to use such novel technologies as multicast routing, IPv6, web caching, etc. The goal of the demonstrations is to test these technologies, as well.

Platform 1 hosted by CNRS, CEG, TCR, MOV, in Limoges, has the largest number of services to be tested. Its main activity will be the demonstration of video on demand services, because this service is expected to be a major traffic volume increaser of the future. The platform will also host services such as entertainment on demand, peer-to-peer networking. This platform will also be the test site of the unified platform created by Moviquity.

Platform 2 hosted by TCL and ING, will demonstrate how a broadband wireless network can provide Internet access to the communities in the rural areas of the Spanish region Castilla-La Mancha. The demonstrator will combine wireless access networking technologies with satellite access and DVB-RCS solutions. The demonstrated services include solutions for distance learning: a typical application of multicast multimedia technologies.

Platform 3 is hosted by Telenor and Nera. Its main purpose is the demonstration of the capabilities of IPv6 technology in the field of multicast services. The main focus is on multicast video streaming.

There will be an additional site, called the test-bed. It is actually a cooperation of the University of Salzburg, the University of Buckingham and the Council for the Central Laboratory of the Research council. The tests focus on all-IPv6 and ad-hoc networking, video multicast and service discovery. After successful validation the result of the layered video multicast service will be demonstrated on Platform 3, as well.

The demonstrations and tests performed on these sites will constitute the basis of the business evaluations of the corresponding services. The reports of these evaluations will be placed into D26.

Contents

1. Introduction

Deliverable D11 relies on two main chapters:

- Chapter 2: "*Services and application*s" is a short introduction of those services that will be tested during the BROADWAN project. This chapter defines "what to test".

  The descriptions are kept short deliberately. A more detailed analysis of the services will be included into D26, since it will contain the business opportunities of the services. A short requirement analysis can also be found in the subsections, respectively. For a more detailed definition of service requirements please refer to D6: "User and service requirements" and D7: "Heterogeneous architecture requirements + environmental issues".

- Chapter 3: "*Demonstration platforms and test-bed*" contains an introduction of the four sites were the demonstrations and the tests will be performed during the BROADWAN project. This chapter basically defines where and how to test.

  The sites include the CNRS platform in Limoges, the "Rural area" platform in Castilla-La Mancha, and the demonstration TEL and NER in Norway, plus the test-bed in the UK, operated by RAL, UBU and USA.

  For each site the infrastructure is presented first, then come the test methods and expected results of those services that are to be demonstrated or tested on the given platform.

The following timetable shows the "*whens*" and "*whos*" of the demonstrations and the tests. The partners will demonstrate and test prototypes and applications on one of the available demo sites.

	DP1	**DP2**	**DP3**	**TB**
VoD, CEG	2005Q1			
VoD, MOV	2005Q1			
VoD, TSR	2005Q1			
EoD, TSR	2004Q4			
peer-to-peer, TSR	2004Q4			
peer-to-peer, MOV	2005Q1			
E-learning, MOV	2005Q2			
E-learning, ING	2005Q2	2005Q1		
E-learning, CNRS	2005Q2			
E-learning, multicast, ING	2005Q2	2005Q1		
Layered video multicast	2005Q3			2005Q2
Multicast, video streaming			2005Q1	
Mobility				2005Q2
Auto configuration				2005Q3
Ad-Hoc networks				2005Q3
Multicast web caching	2005Q3			2005Q2
Performance Parameter Tests				2005Q2

Table 1.1: *Planned timetable*

2. Services and applications

2.1 Unified service platform

2.1.1 Introduction

The Unified Platform which will be used for the deployment of a variety of services, among them Peer-2-Peer, Video on Demand, Distance Learning and Distributed file storage. Services of such a heterogeneous group will have a widespread requirement list of the platform where they will be deployed and reside. The service platform will be build by several modules providing specific platform services which makes it easier to make system updates and also facilitate new service provisioning capabilities as you can collocate new modules for future platform requirements. Some other tasks within BROADWAN is expected to be deployed on a platform and using this kind of general platform that this Unified Platform will provide is of great benefit for both the service provider and the service developer. When we discuss the platform in this document we are only considering the software parts and we leave the actual hardware configuration to be discussed at a later stage since at first a platform should be possible to run at most common server HW architectures.

2.1.2 Analysis

The analysis part of the task aims at defining the required modules of the platform architecture in order to provide the necessary and requested functionality. During the analysis we will review the modules both conceptually and functionally. Finding of the analysis has been placed in another deliverable (D26) and here we will just present a summary.

2.1.3 Platform Functionality

This is a general platform with the purpose of unifying several service categories within the same platform in order to provide a wide variety of services and functionality to the end users with a smooth interaction between services and applications. It is of great benefit for the service providers and service developers since there is no need to learn, maintain, re-implement, and support several platforms, but just one platform to be used for a wide variety of services. This will greatly improve the roll-out time and support and thereby increase the profitability for the involved providers and improve the end-user experience and possibly decrease the end-user cost.

There are several important functions to be addressed by the platform which it should provide to the deployed services.
The platform should provide basic service support functionality for:

- Service user management
- Service deployment
- Resource management
- Management
- Connectivity
- Internal communication (Events)
- Flexible internal structure

Normally you can find a lot more basic functionality placed in the platform, but we have decided to manage a lot of the “basic” services in external services deployed as any kind of deployed service. The different functions are provided via modules in the core of the platform and will be explained in the architecture section.

2.1.4 Platform usage

In order to review and concretise the different functions and the complete functionality of the different parts of the platform we will here describe the platform usage.
Since BROADWAN focuses on networks we want to mention that a network module takes care of all the network connectors and communication that is performed within the platform. It should manage the different aspects of basic connectivity as well as secure connections. It should also be the module that takes the incoming information about the current networks that the platform is connected to. This kind of information can then be propagated to the services deployed in the platform in order to adapt dynamically the QoS, bandwidth usage, encoding and other aspects of their network usage within their service. Nevertheless, we see this module as a service that should be added to the system separately which also ensures a more sound upgradeability.

2.1.5 Supporting Applications

What services will be offered by the platform? That is the main question when we are considering the functionality and the requirements that we have to fulfil from the platform point of view since the platform itself does not provide any services to the end-user. In this deliverable we provide several services that will be running on-top of the platform.
We have to enable the platform to adapt to possible future changes, just like to introduce new services in the future. This is done by having a modular built platform were each and every module of the platform core can be exchanged without any need to change the current services. These core modules should not change their interfaces to the surrounding but only the way they internally work.

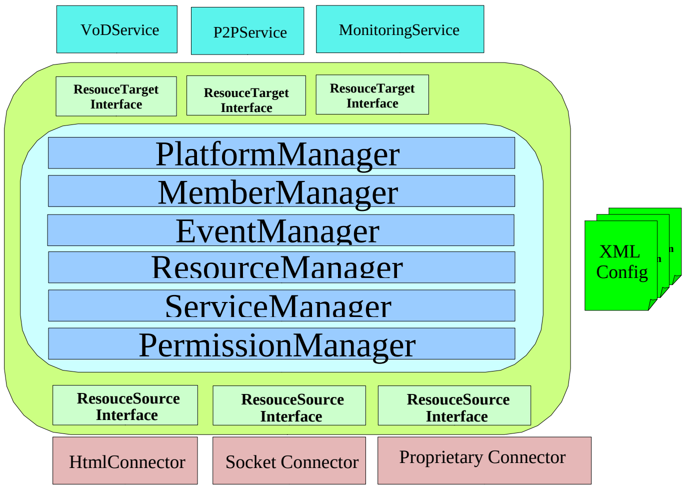


Figure 2.1: *Core modules of the service platform*

If there are new features offered by the platform to the deployed services they will be additional to the previous ones, increasing the possible internal features without changing the current. This is done by defining new services as external service and defining their resource offering, i.e. a deployed service.

2.1.6 Validation

Since the platform will be used for various Service deployments there will be thorough testing of the platform. Nevertheless, there are some parameters that are interesting for us to test, especially the platform characteristics.

2.1.7 Environment Feedback

Environment feedback or *Dynamic configuration* of applications and services is a service that will be deployed in the platform and to be used by several of the services detailed in this deliverable. The reason for describing the service here is to avoid complete repetition of information. As one can see in the service test descriptions there are references to this kind of service, and one should remember that this is a service like any other services deployed on the platform although it might become necessary to see this service as necessary for improved overall functionality for some services.
Dynamic configuration of applications in order to optimise the user experience as well as improving the network performance is of great importance not only for BROADWAN but for all kinds of network based applications and systems. The network architecture that can be used for this kind of supervision system is not as interesting as the application layer network where we can have a client-server, Peer-2-Peer, or other kind of distributed overlay network.
The last decade we have seen different kinds of solutions where agent based middleware was one. The solution we will implement in this project will be based on agents and have these to supervise the network performance and also applications that are based on some kind of distributed architecture, e.g. client-server, P2P. Services that are deployed on a network based on the BROADWAN architecture must be able to adapt dynamically to the current network capabilities since the capacity might change due to weather, link failure, routing problems, application changes, or changes of the available bandwidth. In these cases we have to ensure that the end-user can continue the current services with as little quality decrease as possible, without interruptions or in worst case, restart the services.

2.1.7.1 Functionality

With the growing emphasis on information superiority, any time savings or additional utilization of resources enabled by effective network management becomes increasingly important. Intelligent agents are ideal for assessing information, adapting to dynamic conditions, and predicting future network conditions. In the core of the proposed agent system, the agents share memory and they use majority rule architectures for agent conflict resolution. Different techniques need to be provided for building the agents' shared memory of QoS management solutions and allow the individual agents to share their associations of feedback controls in response to application and user QoS profiles.

The agents will not only be able for low level surveillance such as network performance but also applications and their behaviour. Implementation methodology could differ from one platform and client to another, so it is important to use a generic interface for the agents, platform and the communication. This can be seen in the overview graphics in later sections.

2.1.7.2 Dynamic service configuration

Our purpose of the Agents in our system is to provide information from nodes and parts of the network, application and services to other parts of the network where actions can be taken in order to optimise the service and network usage. The way we do this is not relevant as long as it is convenient, efficient, abundant and feature rich. Agents are the right way to implement this kind of functionality and as the reader will see later in the document, the Agent based systems provide this functionality.

The functionality we need is:

- modules situated in various parts of the network nodes, servers, and PCs
    - These modules should be able to perform measurements
    - Collect application, system, service information and provide this information to other modules
    - Modules should be locatable
    - Should provide information about its tasks, parameters and identifiers
- There should be a way for the different modules in the service network to communicate and exchange information
- Storage facilities
- Monitoring functionality
- Management functionality
- API's for the module interaction, invocation
- Protocols for the communication

The service application or the service provider can use the information collected in order to perform required changes to the service provisioning or service usage. This part should also be a module of the same service network, communicating with the other modules in order to get the required updates as well as the other way around.

As one can see in the above description we are presenting a definition of Agents and it is beneficial to use a already studied implementation and hopefully a standardised way to provide this functionality instead of a proprietary solution. A general Agent description is presented in the next section about the Architecture.

2.2 Entertainment on demand

The broadband connectivity growth shows increasing demand for high quality entertainment over last mile connections such as DSL and digital cable. The growing number of broadband Internet users and digital TV subscribers means that demand for broadband services will climb. Broadband subscribers can now experience high quality television, video-on-demand (VoD), games, streaming media, music downloads, data and content-on-demand services.

For the platform operators, the introduction of new entertainment services for these new broadband customers offers the opportunity of significant incremental revenues on top of their existing voice and data business. The key new drivers for broadband service revenues will be Video-on-Demand and television.
Due to the pervasiveness and growth of broadband, it is now possible to offer a high variety of services over any broadband network. This means that new revenue opportunities exist in any combination of VoD, broadcast television, fast Internet access, streaming media, games, music, or telephony services. However, broadband technology in itself does not equal a compelling service, high value content and additional attractive service features such as interactive applications and personalization are also essential for the success of these applications. The combination of a broadband network, compelling content and service will drive demand for the multi-service mix. In addition to compelling content, service providers need to balance the content protection needs of the content owners with the variety demanded by consumers. A well-designed security framework will allow services providers to make money while protecting content but not hindering consumer choice.
The digital distribution of video games, commonly known as games on demand, is a new way of generating revenue from older game titles. T-Online Germany started Games on Demand on the 18th June 2002. The service is offered for Broadband users, on over the T Online broadband portal (T-Vision)

2.2.1 Service description

Internet based games can be

- Browser based,
- Downloaded one-player,
- Downloaded multi-player,
- PC games with online component,
- Console games with online component.

2.2.1.1 Online games

The categories of on-line games can be broken down to the following two main chapters:

- action (e.g. action, strategic, simulation, adventure, role-playing, and arcade games), where the main factor of the success of the player comes from his or her speed, smartness and manual skills,
- logical (e.g. table games, card games, quizzes, and educational games) where the key factor is cleverness and the educational level of the player.

There are games that belong to both of the above two categories, of course.

Bowser- or web-based online games are typically less than 1 MB in size. The most successful samples are “click+play” type of games. Most of them do not require large bandwidth links, except those for example that are combined with a (real time) video broadcast stream.

2.2.1.2 Online quiz games

A typical example *to demonstrate the capabilities of a broadband wireless network* can be an online quiz game, which is an online counterpart of a TV quiz-show. In this case a live video stream is broadcast to the player, but as a surplus to the TV the player can interactively join the game with a web-based form, where the actual question and possible answers are displayed. (The video is not necessarily live, but it can in no way be a downloadable file, because in this case the player could forward to the answer and find in the downloaded file, ahead of time.)

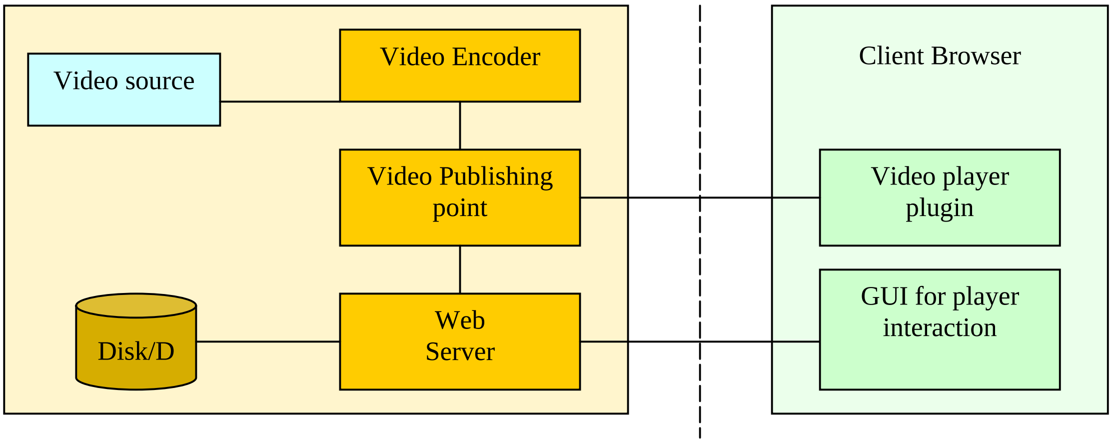


Figure 2.2: *Architecture of an online quiz game with video support*

2.2.2 Network requirements

According to the tables of section 2.3.3, the *bandwidth* requirement of an online game with video content is around

- 512 Kbps on the client uplink side,
- a few Kbps on the client downlink side,
- N*512 Kbps on the server downlink side, where N is the number of connected users in a unicast environment, and 1 in a fully multicast one,
- a few hundred Kbps, on the server uplink side, depending on the number of participating players.

In the case of an online quiz the *network delay* is critical, if the players compete with each other, or even more if the quiz is broadcast on the television, since those with a faster connection will have an unadmittable advantage. Therefore network end-to-end delay should be kept under one second.

2.2.3 Test requirements

The web based game with streaming video support requires a Windows Media server, a web server, a database, all running on Windows 2003 Server. The client side requires a Java 1.4 enabled browser and Windows Media player.

2.3 Video on demand services

Video on demand (VoD) is the future of personal entertainment, offering viewers unprecedented convenience, choice and control. Choosing from a huge selection of top entertainment, subscribers can enjoy VoD any time of the day or night without leaving the comfort of their homes.
It is a growing market for cable operators, with subscription and transactional VoD providing the best short-term opportunity for profitability.
VoD is the technology that allows you to order a Hollywood movie at the exact moment you want to watch it. There are no pre-set start times. VoD also lets you control the movie you are watching just like a VCR would. You have the ability to Fast-forward, Rewind and Pause your movie at any time.
VoD allows users to select and watch video content over a network as part of an interactive television system. There are several types of video on demand systems.

NVoD systems, or Near Video on Demand systems, are systems in which users wanting to watch a film are batched up for the next start time. This is a reasonable model for films, which are in high demand, as the video server can simply distribute the film at short intervals, preferably using multicast techniques. NVoD provides users with a video on demand service, but imposes a short latency delay before the film starts.
Some interactive VoDs allow the user to pause, fast forward, fast rewind, slow forward, slow rewind, jump to previous/future frame etc. In other words: to provide a large subset of VCR functionality. Such systems require more effort on the part of the server, and may also require greater network bandwidth.

It is possible to put video servers on LANs, in which case they can provide very rapid response to users. Video servers can also serve a wider community via a WAN, in which case the responsiveness may be reduced. Nevertheless, it is possible to provide VoD services over a wide area network.
Another interesting application field of VoD services is *on demand advertising*. It is a service where advertiser clients can upload their pre-prepared video of the commercial, and can set a time and date (multiple if necessary) of the delivery. The material can be uploaded to several displays, at different times.
Typical places of displays can be at shopping malls, gas stations and other shopping areas. The video material can be broadcast to the display clients by download or using streaming technology.

2.3.1 Service description

2.3.1.1 Centralized general VoD model

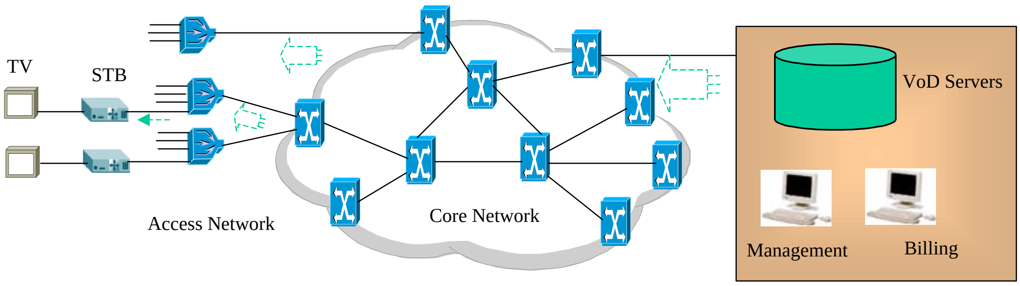


Figure 2.3: *Centralized VoD architecture*

The centralised model is well suited for broadcast TV but not for Video on Demand services.
Indeed, the centralised model is not economically viable and makes the capacity planning process much more complex. Each consumer uses a high bandwidth unicast stream for VoD and this occurs throughout the backbone making congestion more likely.

2.3.1.2 Decentralized model

The decentralised model on the other hand allows the feeding of the VoD servers to be carried out as a delayed process and not in real time. Bandwidth is then consumed by end-users only on the access network
This model must be used to implement VoD services in operator networks.

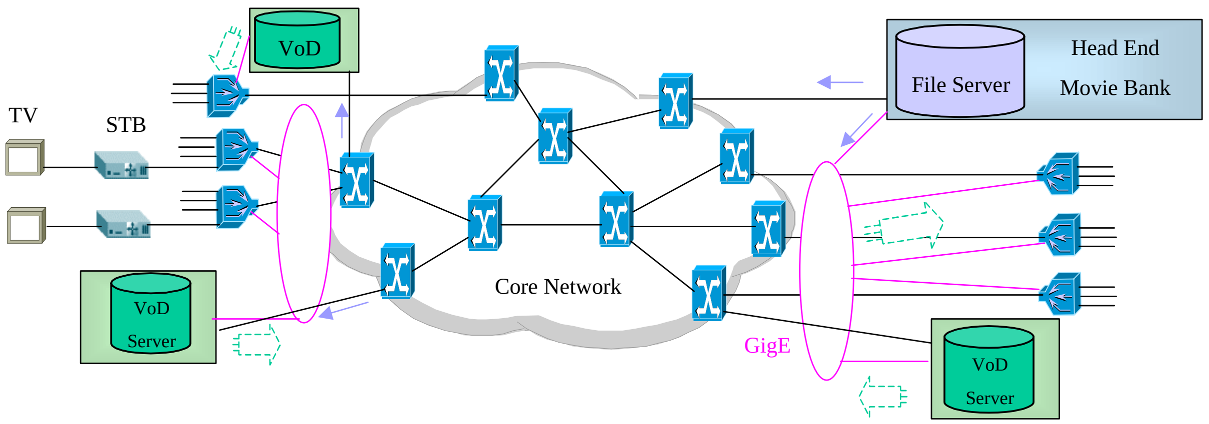


Figure 2.4: *Decentralized VoD architecture*

2.3.1.3 The client side

End users possess client devices to control the video streams and display them.

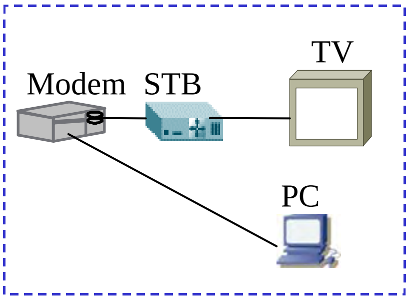


Figure 2.5: *Client side architecture of a VoD system*

They used to watch Video (broadcast TV or DVD) on their TV sets using a remote controller but not on their PC. In order to provide an attractive service, Service Providers are willing to offer VoD in the same platform.
More over, in order to provide a stable high quality service, Providers don't want to cope with problems arising on countless combinations of hardware and software elements found in PCs and must use stable, centrally controlled and secure client devices.

2.3.1.4 Advert on demand

Advertisement on demand is a VoD service, where the demand appears on one place and gets satisfied at another one. Advertising clients can order the place and the time of the appearance of their video material, and the server starts the video stream at the defined time on the defined destination, which is different from place of the customer.
The system can be centralized and decentralized, depending on the balance of available and required bandwidth. If one server cannot service all displays with the necessary stream, the load can de divided among several servers.

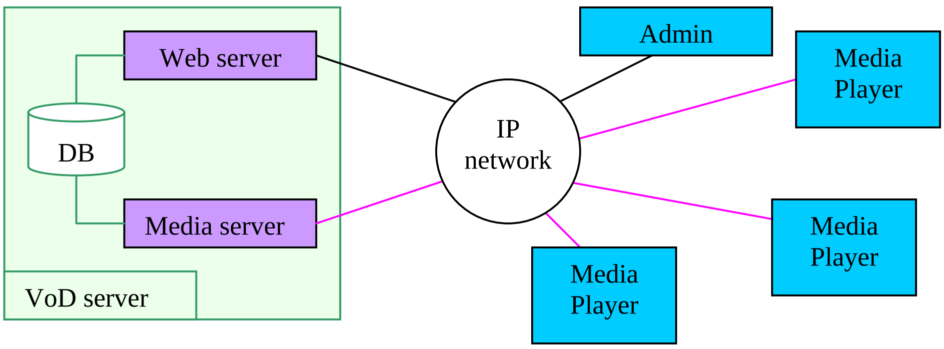


Figure 2.6: *Architecture of the "advert on demand" VoD service*

2.3.2 Service requirements

Video on Demand service requirements can be partitioned in the equipment requirements and the actual service requirements. Starting with the hardware requirements, a VoD server should have the following characteristics:

- Capacity to hold hundreds of GB or more likely TB of Video material all depends on the target use, audience and system features
- Provide simultaneous access to several hundreds (or thousands) of subscribers in real-time, giving each one an appropriate bandwidth, depending on encoding, subscription etc. and is usually in the order of 1.5-6 Mbps

Other consideration that are important when setting up a VoD server are:

- Hierarchical storage scheme
- Scalable Architecture
- Storage Subsystem
- Expanded Storage
- Convenience & quality

For the clients it more depends on the client requirements and service expectation. The clients need to have a multi media client able to connect and reproduce the offered video stream. This require proper (solution dependent) CPU performance, RAM, storage capacity, video card performance, and sufficient network connection to the service provider.
The service in general has the following aspects to take in consideration:

1. Navigation services:
   Embedded in a multimedia environment VoD must be easy to understand and easy to manage; i.e. easy menus for example via graphical in-screen interface, and easy order procedures.
2. Independence of time:
   The user should be able to order any movie at any time.
3. Fast, non-blocking access:
   The system must provide the user with fast response times for interactions and the blocking probability should be below 1% or less depending on QoS profiling.
4. Quality:
   The quality of transmitted videos has to match at least today's standards. Compatibility to future systems (HDTV) has to be considered.
5. Price:
   The price should be in the range of today's alternative products (PPV, video store).
6. Service access control:
   Parental control over movie access is necessary for families with children.
7. Network access security:
   The provider has to guarantee that nobody get unauthorized access to use other user identification or access lines.
8. Protection of individual data and information:
   Information about subscribers has to be protected.

2.3.3 Network requirements

The network requirements of a VoD system are described by the

- bandwidth,
- delay,
- jitter

parameters.
Bandwidth is strongly correlated to the resolution. In a streaming environment the following bandwidth requirements apply as a function of resolution,

Video format	**Compression**		**Resolution**		**Bit-rate**
	Video	**Audio**	**PAL**	**NTSC**	
Video CD (VCD)	MPEG-1	MPEG-1	352 x 288	352 x 240	1.15 Mbps
SVCD	MPEG-2	MPEG-2	480 x 576	480 x 480	1.15 - 2.3 Mbps
ASF	MPEG-4	WMA 9	320 x 240	320 x 240	130 - 680 kbps
			720 x 576	720 x 480	1.15 - 2 Mbps
			1280 x 720	1280 x 720	5 – 8 Mbps
DV	DV AVI	DV AVI	720 x 576	720 x 480	25 Mbps
Real Media	RM	RM	320 x 240	320 x 240	130 - 680 kbps
DVD	MPEG-2	MPEG-2			
(AC3/PCM)	720 x 756	720 x 480	5 – 10 Mbps		
DivX	MPEG-4	MP3	720 x 576	720 x 480	1.15 Mbps

and as a function of video quality.

Video type	**Audio**	**Frame-rate**	**Resolution**	**Bit-rate**
HD quality	6 channel, 48 kHz sampling, 24 bit/sample	24	1440 x 1080	8447 kbps
			1280 x 720	6433 kbps
	Adaptive CBR	29.97	1280 x 720	5137 kbps
	CD quality CBR			5073 kbps
	Multi-channel CBR			5393 kbps

DVD quality	Adaptive CBR		640 x 480	2137 kbps
	CD quality CBR			2073 kbps
	Multi-channel CBR			2393 kbps
Adaptive bit-rate video	Adaptive CBR		320 x 240	1128 kbps
	CD quality CBR			1064 kbps
	Multi-channel CBR			1384 kbps
Video with lot of movements	Adaptive CBR	60	320 x 240	887 kbps
	CD quality CBR			823 kbps
	Multi-channel CBR			1143 kbps
VHS quality	Adaptive CBR	29.97	320 x 240	464kbps
	CD quality CBR			400 kbps
	Multi-channel CBR			720kbps

The required bandwidth is also influenced by the frame rate of the input signal. Typical values for different video standards are displayed in the following table.

Video standard	Frame rate (fps – frame per sec)
PAL	25 fps
Pseudo PAL	29.97/59,94 fps
NTSC	29.97 fps
Pseudo NTSC	25/50 fps
FILM	23.976 fps
SECAM	25 fps

Network delay and jitter values are not as critical as the bandwidth, since they can be controlled by appropriate buffering methods.

2.3.4 Test requirements

The VoD test environment requires a server and at least two client side PCs. For a larger (say 100) number of client, a more robust hardware e.g. a multiprocessor server is necessary with a disk subsystem and a GBitps network adapter card.

2.4 Distance learning in hybrid networks

Distance learning tools are a kind of collaborative tools that can be potentially used by users with different knowledge, from an ICT's (Information and Communication Technologies) point of view. Those users may access this kind of tools or platforms to receive a formation in any matter, and are supposed to have a minimum ICT's knowledge, just enough to manage a program. By means, it is not expectable from them to be able of configuring medium access or to change their PC configuration to access the system, unless it is extremely simple, or it is very well documented or assisted. Therefore the applications, platforms and tools of distance learning should leave from the base of the simplicity, since in principle should be prepared so that any user with some most minimum knowledge of data processing be capable to control it, and that neither the interface neither the difficulty to access the system becomes an impediment or limitation that can limit, disappoint or produce the refusal of the user.

2.4.1 Service description and requirements

The experience of ING after more than five years of development of the IG-Class distance learning platform have carried us to try to achieve the greater simplification of the interface and the applications for the users of distance learning, trying to minimize the learning time of the application, providing web page format and reducing the actions that can carry out the student just to the minimum and necessary when a student assists to a live class, the actions that can carry out are:

- Log-on.
- To Request shift to carry out a question (only available for live classes).
- To expand the zone of view in screen to maximize whether the teacher or the slide.
- To participate in the chat (only available for classes in direct).

IG-Class Student – System main features:
System required by every live or recorded virtual-classroom student

- Fully integrated in standard MS Internet Explorer navigator.
- Live class or programmed and under demand recorded class assistance.
- Reception remotely controlled by teacher without student intervention.
- Supports enlargement of presentation slides and video sequences.
- Interactive multimedia intervention under demand or teacher's will.
- Automatic student's software version update through a Web page.
- Partial disconnection with class reception, without PSTN, GSM, ISDN or any

Internet access connection, therefore without any additional communication cost for student.

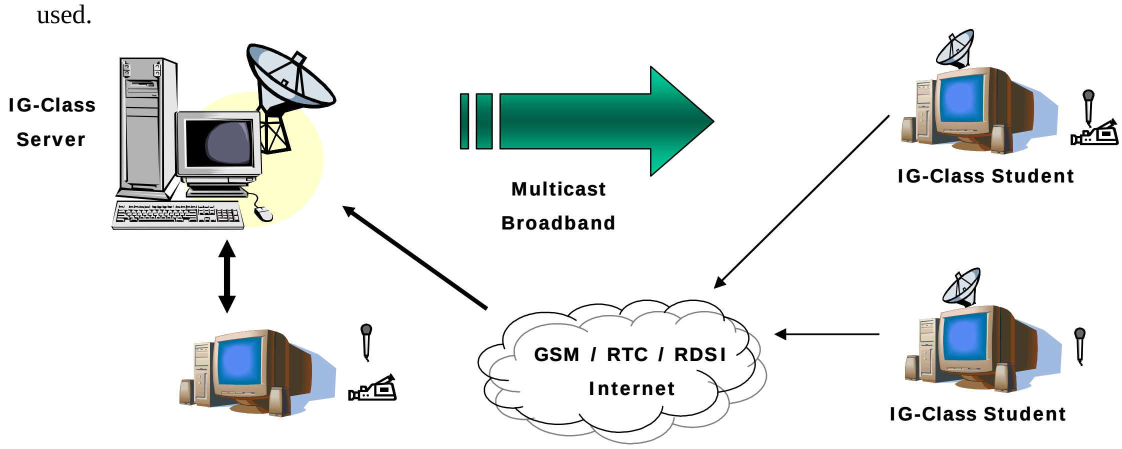
Pre-recorded classes playing from storage devices like CD-ROMs

- Multiple class storage in the same CD-ROM media (up to 4 hours of recorded class).
- Fast forward and direct jump to any part of the class through an integrated historic event list.
- Minimum hardware requirements: basic multimedia and without connectivity.

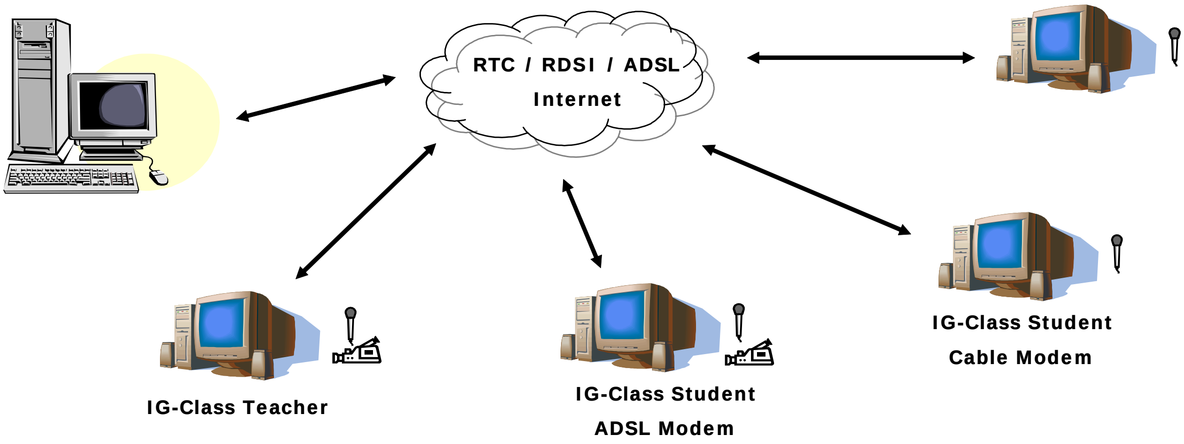
2.4.2 Network requirements

Students access the network from heterogeneous access technologies, which becomes a headache for the network administrator. Two interesting case studies based on different network architecture scenarios can be analyzed:

1. Broadband Multicast IP Operator, as shown in the following figure the broadband multicast platform relies on a satellite delivery system for the downlink (from the server to the end-user) as the one ING implemented for TCL. The challenge for this scenario is that every network equipment must support multicast, otherwise multicast will not flow through the entire network. To rely downlink on just one access technology ease the system dramatically due to, although nowadays multicast networks are not something new, most of the telecommunication platform do not really support multicast traffic indeed, and those really supporting it may not be fully interoperable with other operators. In the scenario shown in the figure, it does not matter how many operators provide the return channel or which technology is used.



2. Broadband Unicast IP Operator, when the service provider can't control the entire network to assure multicast will "flow" from end-to-end through it the only way to provide this service is by streaming unicast from the IG-Class server to each end user. This is the least optimum way to deal with the available bandwidth in the transport network, but it does work.



Any other network architecture in between these two scenarios can be achieved. The more seemed to the first scenario the more efficiently used the network resources of the operator/s.
IG-Class Distance Learning System offers a complete multimedia solution for distance learning using IP networks. With IG-Class geographical barriers will be no longer a problem and students will get the best and most effective education. Ideal for educational centers and corporate employees training.
Integrated by a web based management and information system for courses, students and learning material (asynchronous learning), and a real time multimedia system (synchronous learning).

2.5 Collaborative business environments

2.5.1 Service description

A collaborative multimedia tool (CMT) provides the customer with software for cross-company processing of multimedia content and business documents (henceforth we call them as "documents") as an online service on the Internet, in the context of the existing technical and operational possibilities. The CMT contains multimedia and document management functionalities with corresponding memory capacity on servers in computer center. The computer center are connected to the Internet via a high capacity Internet backbone. The CMT can be accessed as an online service directly from the Internet.
The key features of the CMT are:

1. Secure, cross-company exchange of multimedia content and documents
2. Seamless integration of users, partners
3. High-performance multimedia content and document management system
4. Integration with Windows ("virtual hard disk") and Microsoft Office
5. Tools for tracking tasks and for notification of online provision of the CMT as a hosted service model (therefore no capital commitment or implementation costs)

The CMT improves the distributed processing of multimedia content and documents across company , university or non-business related organization boundaries. All those involved work together transparently and efficiently as if they were in the same room. The use of the CMT offers complete traceability and documentation of all actions.

Figure 2.7: *Collaborative Multimedia Tool*

2.5.2 Service requirements

The CMT provides the following functions:

- Multimedia content and document management
- Project management
- Communication and notifications
- Decentralized administration

The document and multimedia repository of the CMT is used to store any type of document or multimedia content. Users can create various versions and define attributes. Users can also access the CMT as a virtual hard disk directly via WebDAV.

2.5.2.1 Integrated Object Management

Integrated object management in a collaborative environment aims to support document-based communication between business partners. Information exchange must be streamlined according to the business processes taking place in a collaborative environment, support any format for new and existing users and provide a real-time access from different devices within the environment.

Record management includes records management policies for control of records for the collaborative content. Records management facilitates regulation of records within their life cycle in the context of business processes of a collaborative environment.

A record is a piece of any recorded information generated or received by a business unit in a collaborative environment. A record includes the roles of an underlying document in the business processes and relationships with other records.

Records in collaborative environment can be in the form of an electronic and paper-based document or other forms including multimedia content records: scanned images, emails, phone records, audio and video files, faxes, papers, legacy system data, metrics from various sensor devices est.

Records management should facilitate the life cycle stages of a record: creation, storage and maintenance, retention and disposition, and archival preservation. These stages imply the general functional requirements to the record management system according to the DoD 5015.2 certified Records Management Application (RMA) standard:

- *Record series* allow organizing records into groups. Records in a group have a common set of rules controlling records during their life-cycle. Record folders can be created by users inside record series to provide a structure for filing the documents. Any object (records, record series and record folders) must have a unique identifier within the life-cycle of a record management system.

- *Metadata* field(s) belongs to every object in RMA to facilitate search of the object. A metadata field consists of a designator and information to store in a field. Authorized users can edit metadata or RMA must capture metadata automatically.

- *Linking* mechanism allow users to establish document links and events based on these links. A link type consists of a name and identifiers of corresponding documents.
- *Versioning* provides the sequence of modifications of a record. Versions of a document must be retrievable and can be applied to various record series.
- *Cut-off* criteria verify if a record must be cut-off:
  - Time disposition criteria defines a time-based cycle period after which a record must be cut-off,
  - Event disposition criteria describes event-triggered actions,
  - Time-event disposition criteria imply a record to be cut-off only if specific events happen and a time period is finished.

When cut-off criteria for a record are satisfied, records managers determine further actions for dealing with the record: if the record should be accessed, transferred or disposed in accordance with the life cycle.

- *Vital record* status denotes that a record is valuable. A vital record must be inside a vital records folder. RMA must periodically revise if records are vital.
- *Markings* are metadata to be defined to records by authorized users. Security based markings describe access to records. Information markings facilitate records classification.

The set of listed requirements can be adjusted in accordance with needs of particular business processes of a collaborative environment.

Document imaging allows storing and maintaining the electronic copies of off-line documents such as papers and microfilms.

The paper-based documents should be scanned and placed to the storage system. The functionality of the storage system should support documents changing over time, volume and technology.

All the documents stored in the system have to be indexed for providing efficient searching and retrieval to all business units of the collaborative environment.

Retrieval tools can use index or text information within an imaging system to find a document. The retrieved documents must be viewable for any authorized user. A collaborative environment requires a document to be accessible for any business partner's applications and hardware platform.

Automatic conversion to PDF This function enables automatic creation of PDF versions for read-only access and control of the layout. Server-sided engine generates automatically the PDF version (only privileged users can generate or download PDF versions. If a document is "frozen" only the PDF version could be downloaded.

Connection of electronic and paper versions A unique code in PDF documents enables assignment of printed documents to the corresponding electronic version.

Links and document collections Several documents can be grouped together in a document collection with separate version management and access rights.

Document discussions and notes Discussions and notes support the approval and editing process. Privileged users can edit/delete and answer a discussion. Only the owner of a notice can delete it.

Configurable document tasks Processes for approval, release and publication of documents are mapped. Document tasks are redistributable.

2.5.2.2 Project management functionality

Tasks can be created in CMT, assigned to individual users (or user groups) and their current status can be requested. Several tasks can be grouped into milestones and monitored. CMT automatically notifies the users about the task deadline or overdue. Different permission levels ensure the correct structure of the task delegation or milestone creation.

2.5.2.3 Integrated workflow management

The workflow management in a collaborative business environment provides a shared management of tasks and documents structured in a knowledge-based business process. It drives business processes of the customers and allows them to develop, manage and monitor their workflows and integrate their business applications and resources within the environment.

Integrated workflow defines the task infrastructure and tracking, business actors, related resources and document routing. Integrated workflow management should possess customer-focused flexibility and consider that business processes in a collaborative environment are highly distributed.

2.5.2.4 Security requirements

Security in collaborative environment is expected to provide the following (sometimes contradicting with each other) requirements:

- confidentiality of the most critical information;
- integrity protection of information and its processing (creation, transfer and display);
- proved authority of information pieces as well as all and each of a user actions (designated for comprehensive audit);
- good availability of all provided services, reasonable approach procedures and time;
- comprehensive audit: registration and history of all information handling;
- constant on-line and off-line monitoring purposed for quick attack scenario recognition or retaliation (Watchdog processors).

The wide-spread need for IT security provision compelled to join efforts of interested countries for elaboration of commonly recognized criteria to check and evaluate the respective products. The Common Criteria (now acknowledged as ISO/IEC standard 15408) presents requirements for the IT security of a product or system under the distinct categories of functional requirements and assurance requirements. The Common Criteria (CC) functional requirements define desired security behavior. Assurance requirements are the basis for gaining confidence that the claimed security measures are effective and implemented correctly.

2.5.3 Test requirements

With due care for proper development process and defined input for Common Criteria evaluation process a good level of security can be assured. The certification/validation of evaluation results can provide a sound basis for confidence that security measures are appropriate to meet a given threat, and that they are correctly implemented.
The Common Criteria includes an assurance scale (the Evaluation Assurance Levels) that can be applied to help generate different levels of confidence in the security of products. How much confidence is required will be a matter for users to determine, in relation to the value of assets to be protected, the threat environment, and the available budget.

2.6 Peer-to-Peer networks

Peer-to-Peer (p2p) is a very controversial topic where many experts believe that there is not much new in the P2P environment. There is also a lot of confusion about what P2P really is about. The term “peer-to-peer” refers to a class of systems and applications that employ distributed resources to perform a critical function in a decentralised manner. The resources can be computing power, data (storage and content), network bandwidth, and presence (computers, human, and other resources). The critical function can be distributed computing, data/content sharing, communication and collaboration, or platform services. Decentralisation may apply to algorithms, data, and meta-data, or to all of them. This does not eliminate the possibility of centralisation in some parts of the systems and applications if it meets their requirements.
P2P is normally seen as sharing, giving to and obtaining from the peer community. A peer gives some resources and obtains other resources in return. In the case of Napster, it was about offering music files to the rest of the community and getting other music files in return. It could also be donating resources for a good cause, such as searching for extraterrestrial life (SETI@Home), solve biomedical questions of protein-related diseases.
P2P is a way to make use of the vast amounts of computing power, storage from personal computers distributed around the world, and naturally using their network connectivity. Normally the 'P' in P2P is seen as an autonomous entity and that all entities in the network is of the same sort, hence not controlled by the same users. The peers depend on each other for getting information, computing resources, forwarding requests, etc. which are essential for the functioning of the system as a whole and for the benefit of all peers. As a result of the autonomy of peers, they cannot necessarily trust each other and rely completely on the behaviour of other peers,

so issues of scale and redundancy become much more important than in traditional centralised or distributed systems where you normally issue some control mechanism on the access and authority.

2.6.1 Service description

Peer-to-peer systems can be

- pure peer-to-peer,
- pure server/client based,
- using one ore more server components, for administrative purposes

as seen in Figure 2.8.

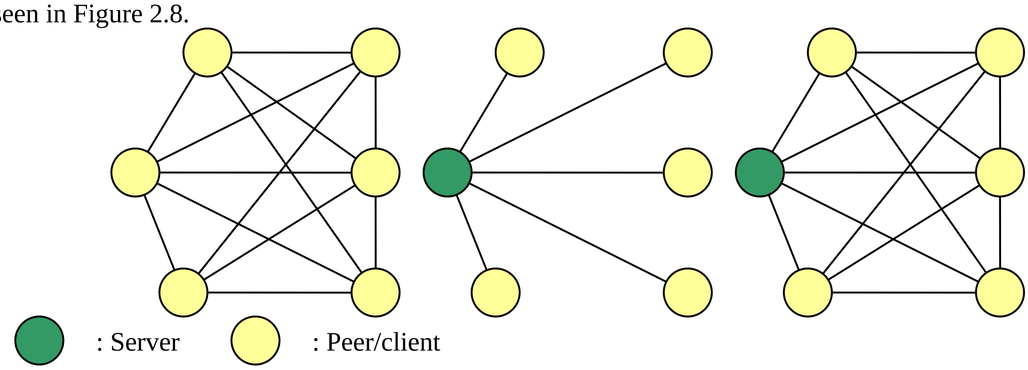


Figure 2.8: *peer-to-peer architecture types*

The third option is probably the optimal, since it gives the power of peer-to-peer connectivity, while adding a server to the system gives the operator the opportunity to have a better control and overview of the system.
The implemented p2p platform is of the third kind. The aim of the service is to demonstrate the capabilities of a peer-to-peer based system.
In a p2p system, usually a peer is not connected to all the others (although they could). Instead they are grouped into user groups. These groups are defined by relationships of the users (friends, family members, club members, etc.). Such a group can be modelled as a normal table (at home or at the office or in a pub), where the members come together and share their views, files and other information.

The prototype
The primarily planned user's group is the group of the home users. A secondary goal is to analyze the feasibility of the Software's functionality to content the needs of organizations.
Here is the list of the main concepts of the system. These are used without further explanations throughout this document. Their introduction describes how the system works from a conceptual point of view, too.
This diagram helps the understanding of the concepts. The details are explained in the document series of the BROADWAN project.

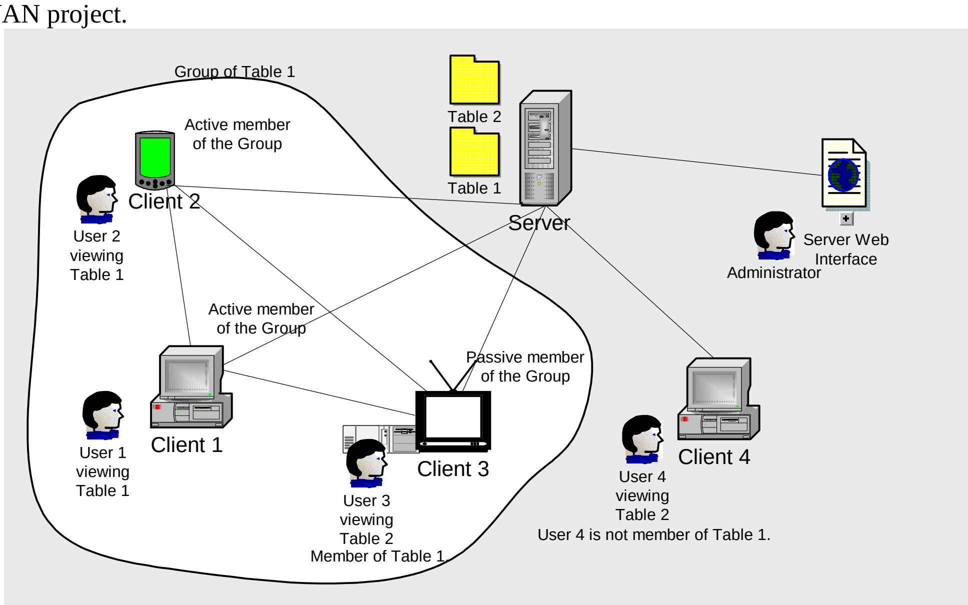


Figure 2.9: *The layout of the p2p platform*

The existence of the Table is important for the implementation of our advanced features. The peer-to-peer file-sharing applications can handle lots of nodes in one virtual group. But the implementation of our whiteboard feature or chat feature is not possible for a big group of nodes. We organize the groups according to the Table memberships of the logged-in Users.
For further details on the architecture of the platform refer to the document series of the BROADWAN project.

2.6.1.1 Ipv6 support

The Software uses Sockets to communicate over the network. The usage of sockets is implemented using the standard Java class InetAddress to identify other network addresses.
The differences in IPv4 and IPv6 are handled internally inside this class (if IPv6 is to be used a subclass of InetAddress called Inet6Address is used, in another case Inet4Address is used - but to our application only the API of InetAddress is presented).
This approach is only possible in Java 1.4 that supports transparent IPv6 handling. If the network layer is used with older version of Java (as in the PDA clients), then the support of IPv6 disappears, but the support of IPv4 remains available.
A user can define in java properties which version of IP he wants to use or which one is preferred with Java 1.4.
Here is an excerpt from Java 1.4 documentation about the support of IPv6:
"Socket, ServerSocket, and DatagramSocket
Due to the object-oriented nature of Java, the address types and storage structures are not exposed at the socket API level, so no new APIs are needed. The existing socket APIs handle both IPv4 and IPv6 traffic.
The selection of which stack to use depends upon the following:
1. The underlying OS support;
2. The user's stack preference property setting.
All supported Ipv6 socket options have an IPv4 counterpart. Thus no new APIs were added to support IPv6 socket options. Instead, the old APIs are overloaded to support both V4 and V6 socket options."
The IP addresses are stored in the database in such a manner that both IPv4 and IPv6 addresses can be used.
The "IP address of the server" command-line parameter for the client program is implemented in such a manner that both IPv4 and IPv6 addresses can be specified as the IP address of the server.

2.6.2 Service requirements

Decentralisation
P2P models question the wisdom of storing and processing data only on centralised servers and accessing the content via request-response protocols. In traditional client- server models, the information is concentrated in centrally located servers and distributed through networks to client computers that act primarily as user interface devices. Such centralised systems are ideal for some applications and tasks. For example, access rights and security are more easily managed in centralised systems. However, the topology of the centralised systems inevitably yields inefficiencies, bottlenecks, and wasted resources.
Scalability
An immediate benefit of decentralisation is improved scalability. Scalability is limited by factors such as the amount of centralised operations (e.g. synchronisation and coordination) that needs to be performed, the amount of state that needs to be maintained, the inherent parallelism an application exhibits, and the programming model that is used to represent the computation.
Anonymity
An important goal of anonymity is to allow people to use systems without concern for legal or other complications. A further goal is to guarantee that censorship of digital content is not possible.
Self-Organization
In cybernetics, self-organization is defined as "a process where the organization (constraint, redundancy) of a system spontaneously increases, i.e., without this increase being controlled by the environment or an encompassing or otherwise external system". In P2P systems, self-organization is needed because of scalability, fault resilience, intermittent connection of resources, and the cost of ownership. P2P systems can scale unpredictably in terms of the number of systems, number of users, and the load.
Cost of Ownership
One of the premises of P2P computing is shared ownership. Shared ownership reduces the cost of owning the systems and the content, and the cost of maintaining them. This is applicable to all classes of P2P systems.
Ad-Hoc Connectivity
The ad-hoc nature of connectivity has a strong effect on all classes of P2P systems. In distributed computing, the parallelised applications cannot be executed on all systems all of the time; some of the systems will be available all of the time, some will be available part of the time, and some will be not be available at all during the scope of the execution. P2P systems and applications in distributed computing need to be aware of this ad-hoc nature

and be able to handle systems joining and withdrawing from the pool of available P2P systems. While in traditional distributed systems, this was an exceptional event, in P2P systems it is considered usual.

Performance

Performance is a significant concern in P2P systems. P2P systems aim to improve performance by aggregating distributed storage capacity (e.g., Napster, Gnutella) and computing cycles (e.g., SETI@Home) of devices spread across a network. Because of the decentralized nature of these models, performance is influenced by three types of resources: processing, storage, and networking.

2.6.3 Test requirements

The p2p platform needs a normal desktop PC (either with Windows or Linux) with J2EE 1.3 and JDK 1.4. The client side also requires a normal desktop PC (either with Windows or Linux) with JDK 1.4 and a graphical display.

2.7 Virtual disk services

To create enterprise-level storage that meets the requirements of many thousands of users, **Storage Networking** concept can be a viable solution.

It is necessary to use Storage Networking instead of regular storage because of the following reasons:

- Bigger amount of data
  The data stored online and transferred between storage and server has increased size, and the amount of data transmitted between server and client PC is large, driven by structured data (text and numeric data) combined with unstructured data (images, audio, and video).
- More sources
  The application must work on several sources of data to satisfy the client transaction. This means there are several online storage units that the server must connect to process the application.
- Single distribution strategy
  The results of the application need to be placed in a central location for access. Generally, this means Internet accessibility.

However, there are two important problems which are needed to be carefully considered:

- Size
  - Wider bandwidth is needed. The connection between the server and storage unit requires a faster data transfer rate. The client/server storage model uses bus technology to connect and a device protocol to communicate, limiting the data transfer to about 10 Mbps (maybe 40 Mbps, tops).
  - The database and supporting online storage currently installed has exceeded its limitations, resulting in lagging requests for data and subsequent unresponsive applications. You may be able to physically store 500 GB on the storage devices; however, it's unlikely the single server will provide sufficient connectivity to service application requests for data in a timely fashion, thereby bringing on the non-linear performance window quite rapidly.

  Solution: Storage networking enables faster data transfers, as well as the capability for servers to access larger data stores through applications and systems that share storage devices and data.
- Data access
  - There are too many users for the supported configuration. The network cannot deliver the user transactions into the server and respond in a timely manner. The server cannot handle the number of transactions submitted, the storage and server components are grid locked in attempt to satisfy requests for data to be read or written to storage.
  - The single distribution strategy needs revisiting. A single distribution strategy can create an information bottleneck at the disembarkation point. It's important to note, however, that a single distribution strategy is only a logical term for placing user data where it is most effectively accessed. It doesn't necessarily mean they are placed in a single physical location.

Solution: With storage networking, user transactions can access data more directly, bypassing the overhead of I/O operations and unnecessary data movement operations to and through the server. Storage networking strategies can address each of these issues and make application strategies like single distribution a successful reality.

Currently, there are two major platforms available for Storage Networking:

1. Network Attached Storage (NAS)
2. Storage Area Networks

NAS (Network Attached Storage) boxes remain thin servers with large storage capacities that function as dedicated I/O file servers. Requests from clients are routed to the NAS server through network file systems that are installed on subscribing application servers. As a result, NAS configurations become dedicated I/O extensions to multiple servers. Usage has given way to popular relational databases (for example, Microsoft SQL/Server and Oracle's relational database products), although these remain problematic given their file orientation.
NAS servers work today in a number of different environments and settings, the most diverse being in storage networking. However, their value continues to focus on data access issues within high-growth environments, as well as how they address particular size challenges found in today's diversity of data types.

Storage Area Network (SAN): An alternative to NAS is the other cornerstone of storage networking, the Storage Area Network (SAN). SANs, like their counterparts in NAS, allow storage to be connected to a network and provide access to multiple clients and servers. The fundamental differences are the methods in which they accomplish this. SANs require their own network to operate, which provides a significant increase to throughput. SANs also provide direct I/O access from the devices connected to the network, thus providing yet another fundamental shift in distributing the I/O workload among applications.
SANs are constructed with many new components from the storage network. The foundation is the FC switch, which provides the physical connection that allows "any-to-any" communications within the fabric. SAN switches provide the hardware and software foundations needed to facilitate the network-the hardware itself being composed of ports that permit the connection of FC-based devices, such as storage arrays and servers.
To participate in the storage, network servers require a special host adapter, similar to a network adapter known as an FC Host Bus Adapter (HBA). The HBAs supply the storage network drivers that allow the server to communicate with the switch and ultimately log in and communicate with storage devices.

Finally, storage devices used with the fabric must be FC-compliant devices (that is, they must speak FC to communicate with the network). As with Ethernet networks, early users of SANs required the use of bridges and routers to allow traditional storage devices (such as tape drives) to participate within the fabric. These devices translated FC protocols to SCSI bus level protocols, basically breaking frames down into bus segments so a SCSI-level device (such as a tape drive) could be connected to the FC fabric network.

2.7.1 Service description

The aim of virtual disk services is to provide individualized data storage for the end-customer.
The benefits of these services are two-fold:

- additional storage capacity,
- secure, centralized storage.

Additional storage is relevant in thin-client environments, where the client is physically not capable to hold the required amount of data. The networked storage resources will extend in this case the capacity of the client. Ideally, such solution is implemented in a transparent manner, i.e. the user is not aware where his/her data is stored and can use both resources in a seamless manner. Nevertheless, although less comfortable a user-controlled centralized storage offers still significant benefits.
Secure, centralized storage is relevant for all user and client types. In this case the focus is not on operational usage of the resources, but more as a backup and / or archive solution. A secure, centralized (off-site) storage would significantly extend the availability of the client data as it offers protection from all major factors which can cause data loss:

- hardware failure or malfunctions,
- software problems incl. malicious code (virus, Trojan etc.),
- user errors such as accidental deleting of data.

It has to be emphasized, that security of an off-site storage can be provided only with appropriate procedures which ensure and control the transfer of data to and from to the secure storage.

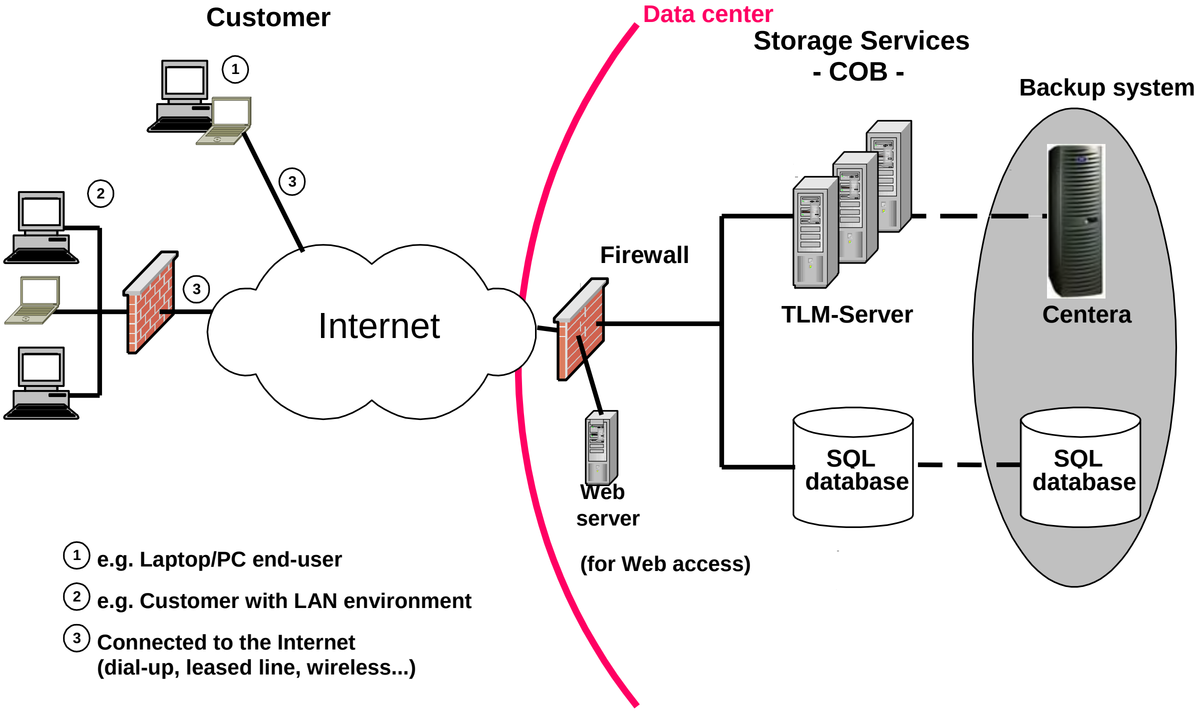


Figure 2.10: *Architecture of the Data Center*

2.7.2 Service requirements

Service requirements are generally depending on the applications and their interactions with the human user. Applications, which actively interact with the user, are relying on steady, reliable and predictable data flow. An example for such an application would be streaming media.
Following service parameter are relevant for this type of service

- bandwidth,
- error rate,
- bandwidth predictability (how likely that the required bandwidth is available through the entire connection (e.g. video session)),

Applications, which work in the background (e.g. a backup application), are less susceptible for the above factors. However, here is relevant how far these applications interfere with the usability of client, i.e. the degree of additional load which the application generates. This can be measured through:

- data throughput generated
- bandwidth requirements
- loss rate

2.7.3 Network requirements

NAS Network Requirements

In case of NAS Enterprise Network Deployment, many corporate applications must be accessed through the data center. The data center must support multiple user access, applications, and data, its requirements for network segmentation become comprehensive as well as external access to the Internet, and dedicated leased lines.
The ERP system is processed and accessed through the UNIX servers within the data center. The information is contained in every system in the data center, the mainframe, the UNIX servers, and the web servers.
Two solutions are possible. First, is the use of NAS devices for the collection and storage of web data. Second, is a large NAS device to hold the sales data that is accessed by sales personnel in the field. An additional important installation would be the use of NAS devices to hold large data base. Each of these solutions can be placed within the data center for general access.
Given that the data center connects to the network backbone and contains a secondary network backbone within the data center, the bandwidth to handle multiple user access can be configured with high-speed network resources and devices. Switches which are used in each segment of the data center provide additional flexibility in the placement of NAS devices given their I/O responsibility.

SAN network requirements
SAN is a network and, as such, its main task is to provide communications among devices attached to it. It does so through the FC standard protocol, denoted by the FC standard and relative T-11 standards committee that maintains the protocol. The FC protocol offers a layered approach to communications similar to TCP/IP, but through a smaller set of processing layers. The FC switch device can provide any-to-any connectivity matrix that supports communications between the devices.

Due to a Fibre Channel, SAN can provide a way for existing protocols to be encapsulated within the communications. This is especially valuable, as the SCSI commands for disk operations do not have to change. Consequently, disk and driver operations can operate within the FC protocol unchanged.

However, it is important to note that in developing the SAN network, additional components are required beyond the FC switch for both the server and the storage devices.

2.7.4 Test requirements

Generally, such testing in this case would mean to test the interaction between network and clients in different situations. Obviously, for practical reasons such tests can be best performed in a simulation environment, where a realistic network configuration can be tested with a multiple of concurrent operating clients. The tools, methodology and practice of such simulations are sufficiently documented in relevant publications. More complex task is to find a) a realistic input for the simulation, and b) benchmarking options to support the qualitative evaluation of the system.
Input values consist of:

- Network topology
- Applications and their behaviour
- User number and behaviour

Benchmarking can be either based on existing similar applications, or relevant to different input scenarios.

2.8 Multicast services

2.8.1 IMPPS – Multicast Web Caching

Multicast Web Caching service will be validated using the Intelligent Multicast Push and Proxy Service (IMPPS) multicast Web cache software. A predecessor of the IMPPS software has been developed and successfully validated in the IST-Project EMBRACE.

IMPPS is a distributed system which acts as an intermediary proxy for HTTP based traffic, with the goal to accelerate Web access and optimize broadcast channel bandwidth usage for interactive Web services. To achieve this, it acts as a distributed HTTP proxy service; if a web document is not available in the local IMPPS instance, the local IMPPS instance may request it from a server IMPPS instance. This instance fetches the requested document, e.g. from the origin server, and returns it to the IMPPS client. By utilizing reliable multicast over IPv6, requested documents may additionally be transmitted to additional IMPPS clients at the same time, which store them in their local cache and are able to return them immediately should users request them, without re-transmission over the broadcast link.

To prevent caching of non-relevant content and thus optimize cache hit ratio and required cache size, IMPPS uses advanced filtering mechanisms. Cache preloading techniques are used to update frequently-used Web sites during off-peak hours, utilizing otherwise unused bandwidth. This preloading mechanism employs rateless error correction codes and layered multicast to provide maximal system scalability.

When used on links with high latency such as satellite connections, IMPPS is able to use HTTP prefetching techniques. Here, the server fetches objects embedded in a Web site (such as images or style sheets) and linked documents in the background to reduce the latency perceived by the users.

IMPPS allows a convergence between IPv4 and IPv6 at service level. IMPPS fully supports IPv4 and IPv6 for unicast and multicast connections. While IPv4 multicast routing is already deployed in most operating systems, IPv6 multicast routing functionality is often missing. IMPPS can be used to provide support for both protocols in parallel, at service level, and to allow for a smooth migration path towards a fully-IPv6 network.

Multicast Web Caching stores peer content locally. As a result, significantly more often-requested content is locally available and does not need to be fetched from the Internet, which results in a notably lower latency during browsing the Web, and which provides a better experience for users. This is also an important characteristic because latency, as opposed to bandwidth, can not usually be reduced by upgrading the transmission hardware. Moreover, less bandwidth is required because of the local availability of many Web documents, reducing the number of required transmissions; thus, providers can support more users with the same link capacity.

2.8.1.1 Service requirements

IMPPS acts like a HTTP proxy. Thus, it requires access to HTTP servers in order to work properly. This access can be provided either directly via a link to the public Internet, or indirectly using HTTP proxy servers. In addition, IMPPS requires HTTP enabled client applications for service validation.

2.8.1.2 Network requirements

IMPPS requires an IPv6 enabled network with IPv6 multicast routing support. In order to increase the bandwidth efficiency, IMPPS clients and the IMPPS server need to be connected to a broadcast network.

2.8.1.3 Test requirements

For testing IMPPS in a test-bed setup, at least one IMPPS server, two IMPPS clients, one HTTP server and two HTTP clients are required.

2.8.2 Adaptive Video Multicast

2.8.2.1 Service description

Streaming of video over IP multicast is already widely in use today, such as for distributing music video clips or movie trailers, and for best-effort live real-time Internet video-conferencing.

However, the current services usually operate over unicast, end-to-end connections using the HTTP or RTP protocol. This quickly becomes very inefficient for larger numbers of customers, and it requires expensive servers and high access bandwidth. It therefore hinders the adoption of many new services such as Video-On-Demand distribution, Internet TV or tele-learning and tele-conferencing; for the latter, efficient usage of multicast and broadcast network facilities, as well as QoS provisions (where available) are important.

An issue of special interest within the FWA links used in the BROADWAN project is the behaviour of multicast distribution over links with limited or quickly varying quality and bandwidth. The adaptive video multicast service allows the transmission of MPEG-2 / Motion JPEG2000 video over IP multicast in several layers, which may either be re-combined with the full video quality at the receiver if all video data can be received, or at least with reasonable diminished quality if the link to the customer does not have sufficient quality.

This technique is a technology preview, as today this is a topic of current research activities in progress.

2.8.2.2 Service requirements

The following services will be required for practical usage by clients. They are optional for the test-bed environment, but may be integrated should they become available:

- Metadata exchange service (optional)
- Digital Rights Management scheme (optional)
- Encryption, authentication of users (optional)
- Network information service, providing current network information (see also network requirements, below; optional)

2.8.2.3 Network requirements

- Basic IPv6 network connectivity
- Multicast routing support
- Quality-of-Service capabilities (optional)

- Provider for network meta-information (e.g. nominal bit-rate of uplink connection; optional)

2.8.2.4 Test requirements

Test platform with 1 server hosts and 2 or more client hosts

2.9 Service discovery

2.9.1 Service description

Service discovery forms an important part of all novel services, as in the BROADWAN project, discovery of network services over wireless (partially unidirectional) links is of high importance. Service discovery in this context does not include discovery of specific devices located in the network, but concentrates on providing information on what session or application layer services are available in the current location. Service discovery thus relies on an already successfully established network connection.

The spectrum of usage of service discovery is wide spread. In contrast to statically available services, which are publicly known to be available at constant resource points, service discovery mechanisms allow to dynamically announce and even shift on-going services. For example, mobile users want to be informed about global but also geographic localised services available as soon as the mobile equipment is turned on and ad-hoc or auto-configuration is finished. Users located in FWA areas utilising non-mobile equipment may be interested in global services offered by the respective service provider(s).

Service discovery consists of three interconnected parts that need to be analysed separately: the service description meta-information, the overall service discovery architecture and the service discovery protocol that is used to exchange the meta-information within the architecture.

Several different solutions for service discovery exist today, most of them defined by the IETF. Grouped into main usage areas, service discovery is implemented or planned the following way:

- Internet applications (VoIP, etc): SIP/SAP and SDP
- Digital Television (DVB): ESG using PSI/SI
- DVB-H: ESG using PSI/SI and SDP(ng)/XML over FLUTE

2.9.2 Service requirements

Service discovery does not require any special services. However, as soon as the decision is due on what (combination of) currently available and established service discovery meta-data formats and protocols is used, together with custom extensions, it has to be assured that those basic services required are available.

2.9.3 Network requirements

As basic service discovery information is distributed using multicast service, and all services are intended to be deployed in an IPv6-only network, a multicast-enabled IPv6 network is required for the service discovery service.

2.9.4 Test requirements

To perform tests with service discovery service, at least one of the novel multicast services should be installed in a test-bed / demonstrator platform, and correct service information meta-data has to be assembled in order to announce the service. To demonstrate the efficiency, a couple of users within different connection-“areas” (like mobile, FWA, etc) should be available.

3. Demonstration platforms and test-bed

Chapter 3 introduces the four demonstration and test sites where the validation of the services will take place. For each site first the platform description is introduced, then come the list of test procedures and the expected results for each service that is going to be introduced on that platform.

3.1 Demonstration platform 1: Limoges

The LMDS demonstration platform located in Limoges has been developed in a previous RNRT (Réseau National de Recherche en Télécommunications) project : ERASME, implying various french partners (industry, operator and academia). Through the experimentation of broadband services (like VoD, e-learning, videoconference and database acces for students), the prime objective of ERASME was to study and model the data flows generated in order to define wireless acces networks able to support them. The second objective was to overcome technological bolts (hardware or software) related to standards and interfaces compatibility, and deployment conditions. The experimental system has been established in Limoges and includes wireless equipments giving access to broadband services in the 40.5-43.5 GHz frequency band

3.1.1 Platform Description

The overall demonstration platform located in Limoges is shown on figures 1 and 2.

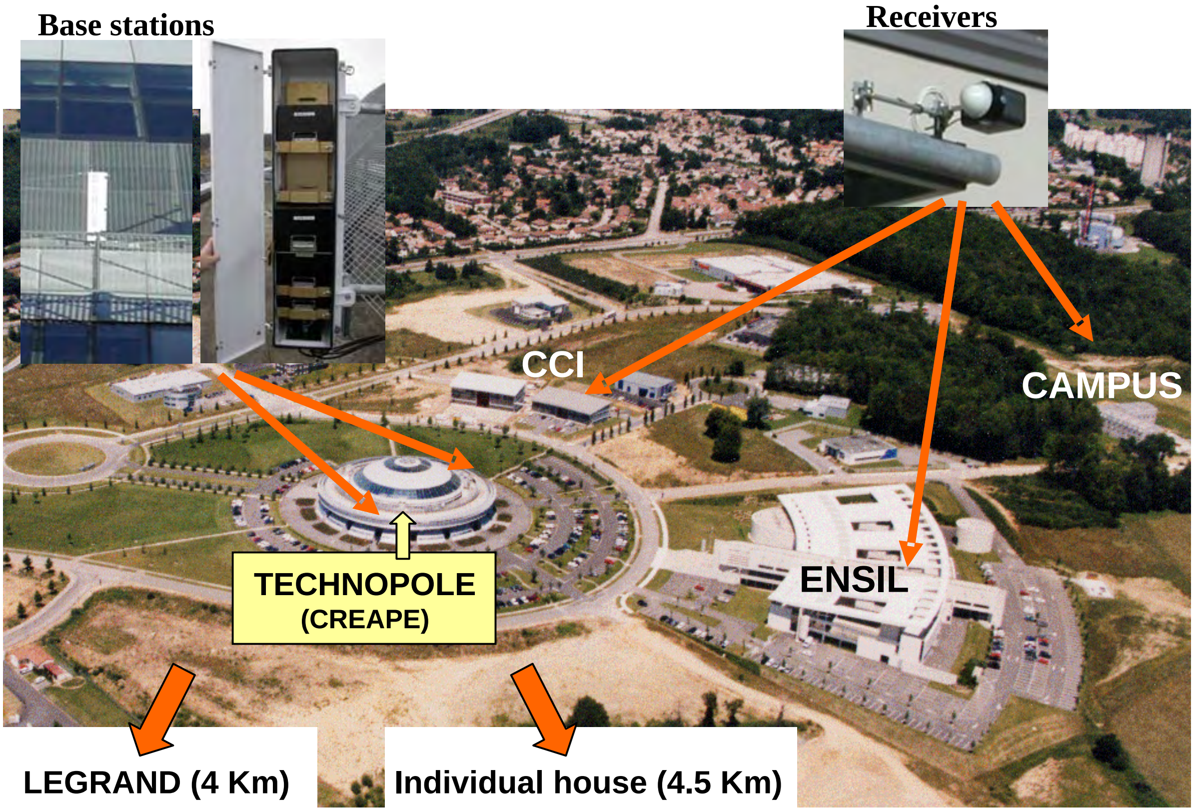


Figure 3.1: *Aerial view of Limoges platform*

Two base stations are situated on the TECHNOPOLE, and cover two sectors (10 ODU). The first sector is constituted by ENSIL, CCI and the CAMPUS, and the second sector corresponds to LEGRAND society at 4 km and an individual house at 4.5 km. One of the ODU in the first sector is located on the TECHNOPOLE (for CREAPE test-link) as shown in Figure 3.2.

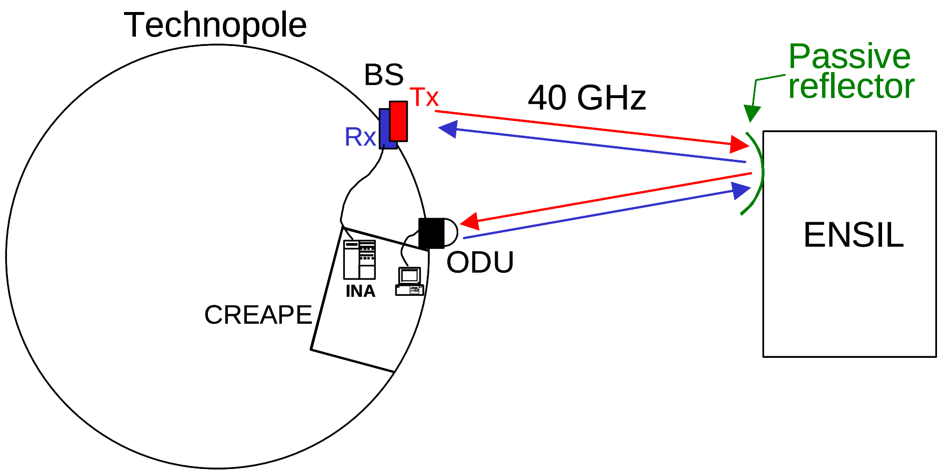


Figure 3.2: *CREAPE test link*

3.1.1.1 BFWA 42GHz platform

The platform network is described on Figure 2.1. The standard used is DVB EN 301 199 (see Deliverable 3, page 36). The platform uses one downstream channel of 33 MHz bandwidth (with a theoretical rate of 34 Mbps) to synchronise 8 upstream channels of 2 MHz bandwidth (with a theoretical rate of 8x2 Mbps). Each upstream channel is divided into 3ms time periods, and one period corresponds to 18 time slots which can be used by different users through the technique of TDMA (Time Division Multiple Access).
The Standard defines three available access modes: Fixed Bit Rate, reservation or contention access. In this case, only the Fixed Bit Rate access is implemented: One (or more) time slot is allocated to each user, every 3ms period (theoretical NIU rate: 128 Kbps if one slot is allocated).
In real cases tested, the NIUs used on the BFWA platform present two major problems:

- data pump saturation which causes a reboot of the NIU. For this reason, the downlink rate obtained for one NIU is 2 Mbps and the uplink rate is 800 Kbps (with random reboot).
- multicast address not supported (MAC problem).

3.1.1.2 Network equipments

Actually, the core network router is a CISCO 3550-12T, a layer 3 switch, with ten 10/100/1000 Mbits ports and 2 GBIC-based Gigabit Ethernet ports.

Internet link (100 Mbps) via RENATER (Réseau National de Télécommunications pour la Technologie, l'Enseignement et la Recherche).
RENATER network provides high rate Internet access for educational establishments, and different centers having activity in the field of research and technology.

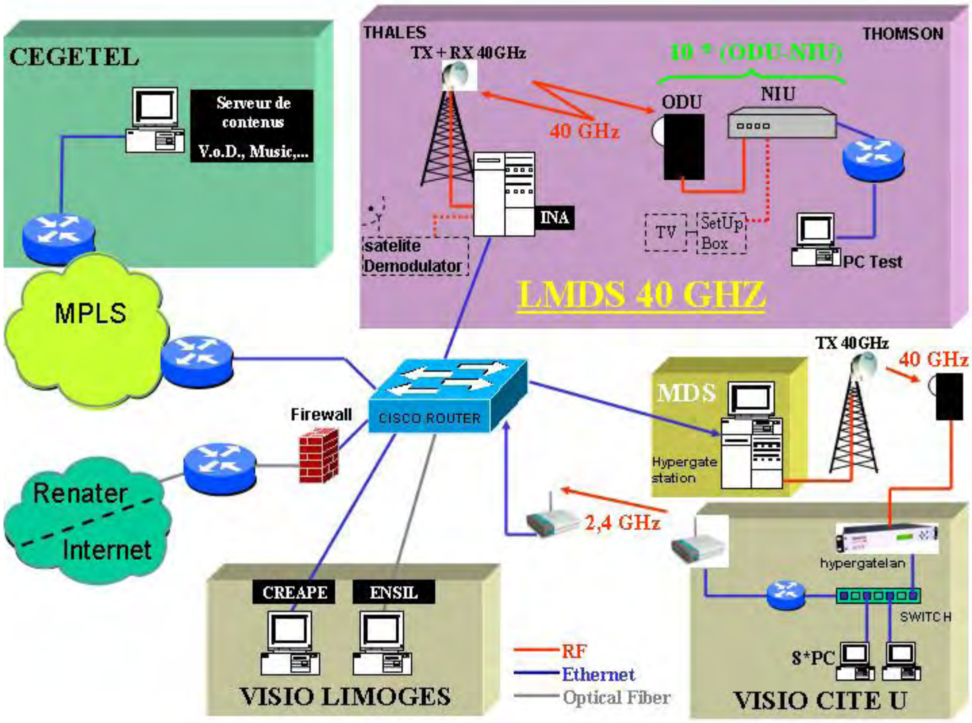


Figure 3.3: *Limoges platform network*

3.1.1.3 MDS Hypergate system

To overcome the NIUs problems described previously, the actual videoconference system is tested by MDS server (hypergate station) using a UniDirectional Link Routing (UDLR), with a 42 GHz downlink at 35 Mbps and a WiFi (2,4 GHz) uplink at 6/8 Mbps (Figure 3.4).

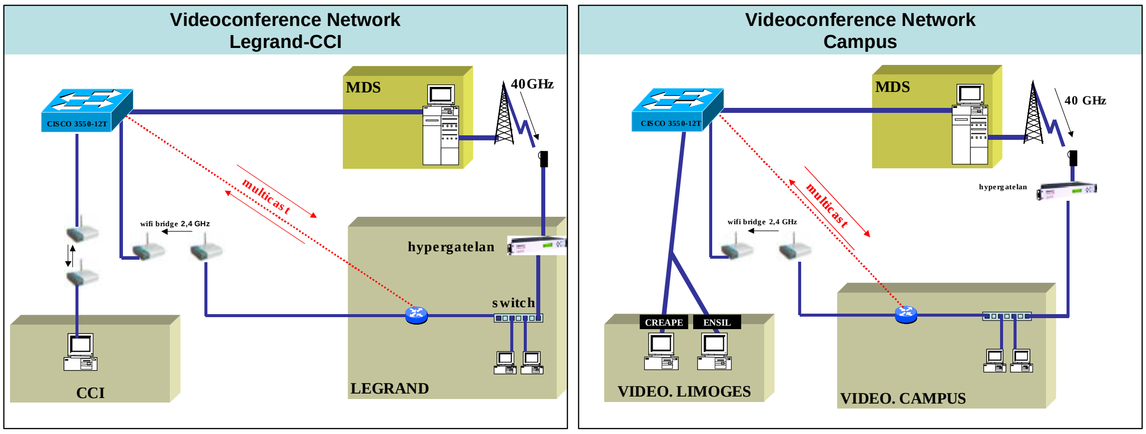


Figure 3.4: *MDS system map*

3.1.1.4 Videoconference

To test the videoconference, one "teacher-room" is located in CREAPE, and another in ENSIL. The equipment for each is listed below:
PC (P3 1 GHz), Video camera Sony EVID 31, Ebeam whiteboard, video projector, …
End-user videoconference equipment:

Campus: 8 PC (P3 1 GHz), 8 web cams under Windows XP.
Legrand: 2 Sony EVID 31, 2 PC.

3.1.1.5 Video on Demand architecture

Access to VoD server based in Paris via CEGETEL MPLS network at 20Mbps.

3.1.1.6 Platform evolution

IPv6 migration

The integrated Cisco (Catalyst 3550-12T) IOS software actually support Ipv4. Upgrading the IOS to IPv6 will not be possible. So, the IPv6 routing will be ensuring by a PC under Linux or by a new router (CISCO 36xx.).

RENATER could interconnect the platform with the IPv6 Internet. A sub-network will be assigned.

MDS Hypergate station server does not support IPv6 actually. IPv6 encapsulation card will be available in future. IPv6 in IPv4 tunnels will be set up to overcome this problem.

Services known to work in Ipv6 technologies:

- Multicast videoconference
- VoD with Microsoft Multimedia Services since Windows 2003 Server
- RENATER IPv6 interconnection : DNS, IPv6 to IPv4 gateway

To test new IPv6 services, an IPv6 only network has to be set-up in a first time. Once the IPv6 network is operational, dual stack environment will be installed. To interoperate, the router must be configured in a 6to4 gateway.

Heterogeneous networking

In order to test services on hybrid networks (LMDS + WiFi), WiFi access could be provided to students living in the campus (3 floors building). Then 3 WiFi access points have to be deployed to give access on each floor as described on figure 5). In this case, a solution to authenticate the wireless students has to be set-up. A survey must be done to know how many students may me interested to participate.

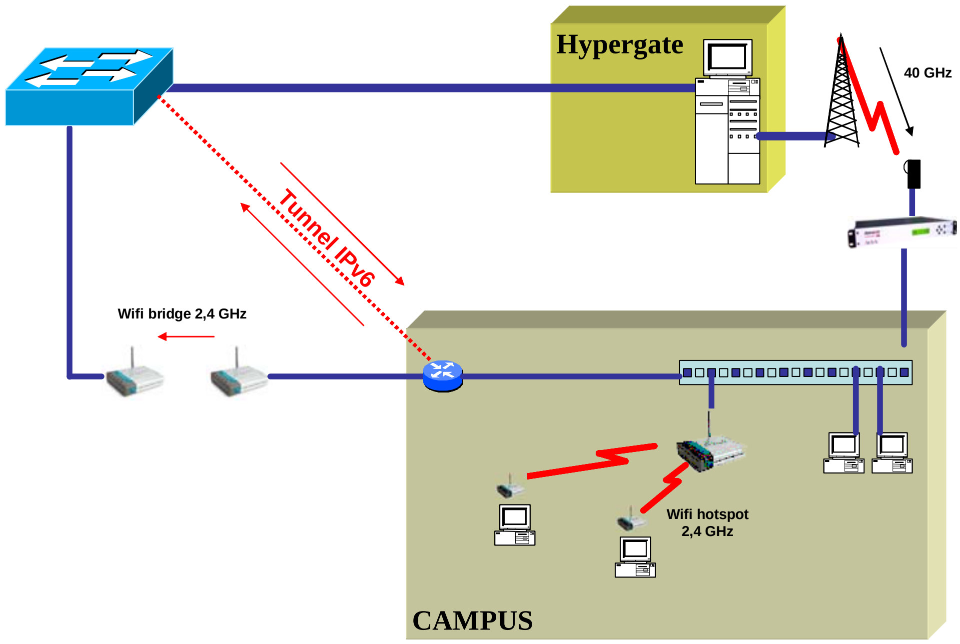


Figure 3.5: *Heterogeneous networking*

3.1.1.7 QoS measurements

Proper QoS management is a dominant requirement in novel network. ISPs have to evaluate the network congestion and allow new services to work perfectly. Today, many new services are offered to subscribers like Telephony over IP, Video on Demand and videoconference.
QoS can be determined by connectivity, delay, jitter, lost packets and bandwidth usage:

- Connectivity is important for every type of application (maybe except mail, etc.)
- Delay reflects the quality as seen by the user and is important for real time apps
- Jitter is due to router queuing buffers. It reflects the network stability. A stream need to have a constant jitter to prevent software buffer dump
- Packet loss is important for every type of usage. They are often due to network congestion
- Bandwidth usage is required to evaluate network congestion.

Service examples:

- VoIP : delay < 100ms, jitter < 50ms and moderated lost packets
- VoD : constant delay, moderated lost packets
- Videoconference : delay < 100ms, jitter < 50ms and significant symmetrical flow

To measure QoS on a network, 3 possibilities could be isolated:

- Cisco Service Assurance Agent is integrated in IOS. CISCO routers can measure: Round Trip Time with UDP packets, jitter, lost packets and several other performances. The results are collected via SNMP. Synchronisation is based on NTP.
- Transparent commercial probe like IPANEMA which are synchronised GPS to and can be placed everywhere. It acts as a transparent bridge.
- Probe based on free software (ex: NIMI, RUDE, Netmate, ...).

3.1.2 Unified platform services

Dynamic Service configuration
The Dynamic Service Configuration which is a part of the demonstrations to be verified on the Demonstration Networks, particularly in Limoges. We will aim to show the functionality of the feedback system that is implemented as a service on the Unified Service Platform using Agents in order to collect and make decisions regarding the configuration of a specified service during its execution. It has been mentioned in several of the service descriptions earlier in the document and will be resumed shortly here.
Description
The Feedback Service can be implemented in many way and we will go for more complete Agent solution that in self is working in a Client-Server fashion where some Agents are specialised on collecting information where others are retrieving this information since their speciality is reading information with this specific contents. The Agent that is working towards the server will provide the Distance Learning Service with information about how the service is performing at any given moment. Then the Service can alter its execution and service for the current situation reflected in the feedback from the Network Feedback Service delivered by an Agent. It is here we will have a Dynamic Service Configuration that can alter the configuration for Bandwidth craving applications to be more responsible and let other applications take part of the current up- and downlink capacity.

Active feedback from the end-user applications, the network, platform or the service will render in new service input which should be used according to configuration of the specific service. This can improve the end-user experience or improve specific network conditions which are highly desirable and the way to implement the actual feedback mechanism is discussed in the section of the Agent Usage for automatic configuration updates.

The feedback coming in to the service will be treated according to configured rules where the transmission bit rate could be decreased if the network is being heavily congested in order to improve the end-user quality although that it also has decreased. This could be done by lower the resolution, frame rate, or frame size. This is of course a drawback for the end user, but is not necessarily a bad thing. It is more though as improving the network conditions in favour of other services that might have priority to the network usage.

The other way around is naturally also possible. If there is low usage of the network thus high on available network resources, the surveillance service can treat the feedback indicating the available bandwidth with increasing the throughput for the end-users.

Dynamic configuration of applications in order to optimise the user experience as well as improving the network performance is of great importance not only for BROADWAN but for all kinds of network based applications and systems. The network architecture that can be used for this kind of supervision system is not as interesting as the application layer network where we can have a client-server, Peer-2-Peer, or other kind of distributed overlay network.

The last decay we have seen different kinds of solutions where agent based middle-ware is one. The solution we will be implementing in this project will be based on agents and have these to supervise the network performance and also applications that are based on some kind of distributed architecture, e.g. client-server, P2P. Services that are deployed on a network based on the BROADWAN architecture must be able to adapt dynamically to the current network capabilities since the capacity might change due to weather, link failure, routing problems, application changes, or changes of the available bandwidth. In these cases we have to ensure that the end-user can continue the current services with as little quality decrease as possible, without interruptions or in worst case, restart the services.

Functionality
With the growing emphasis on information superiority, any time savings or additional utilization of resources enabled by effective network management becomes increasingly important. Intelligent agents are ideal for assessing information, adapting to dynamic conditions, and predicting future network conditions. In the core of the proposed agent system, the agents shared memory and they use majority rule architectures for agent conflict resolution. Different techniques need to be provided for building the agents' shared memory of QoS management solutions and allow the individual agents to share their associations of feedback controls in response to application and user QoS profiles.

The agents will not only be able for low level surveillance such as network performance but also applications and their behaviour. Implementation methodology could differ from one platform and client to another, so it is important to have a generic interface for the agents, platform and the communication. This can be seen in the overview graphics in later sections.

Dynamic service configuration
Our purpose of the Agents in our system is to provide the possibility to propagate information from nodes and parts of the network, application and services to other parts of the network where actions can be taken in order to optimise the service and network usage. The way we do this is not relevant as long as it is convenient, efficient, abundant, and feature rich. Agents are a really good way to implement this kind of functionality and as the reader will see later in the document, the Agent based systems provide the following functionality.

The functionality we need is:

- modules situated in various parts of the network nodes, servers, and PCs
    - These modules should be able to perform measurements
    - Collect application, system, service information and provide this information to other modules
    - Modules should be locatable
    - Should provide information about its tasks, parameters and identifiers
- There should be a way for the different modules in the service network to communicate and exchange information
- Storage facilities
- Monitoring functionality
- Management functionality
- API's for the module interaction, invocation
- Protocols for the communication

The service application or the service provider can use the information collected in order to perform required changes to the service provisioning or service usage. This part should also be a module of the same service network, communicating with the other modules in order to get the required updates as well as the other way around.

Validation of service
What we plan to do is a validation of this kind of system for dynamic service configuration for Service platforms in general and particularly for high performance applications and services such as Video on Demand (VoD), Distance Learning, and as the section header says to Capacity Demanding Services.

Agent based services has been around for a while but the application area that we are applying it to and the way we are going to use the service is not that usual. The big reason to use this kind of service provisioning for a platform is mostly due to the hybrid network and “sensibility” of wireless networks to be used for heavy broadband communication with most probably saturated links. Not all traffic on these links are sensible to quality decrease but some of them are and if we can provide a way to improve or maintain quality on these by decreasing quality for others we have improved the end user experience.

We will be reviewing the Dynamic Service Configuration from different viewpoints that are interesting for the services that is using this kind of service when deployed at the Unified Service Platform such as:

- Configuration update delay
- Robustness
- System and Network overhead
- Service user friendliness
- Possible application areas

3.1.3 Entertainment on demand

Tests and Means

The online quiz game is a web based client-server service, therefore the demonstration will use a server with Windows media services and a web server. The clients will run the game (a Java applet) in their browsers, as well as the video content.

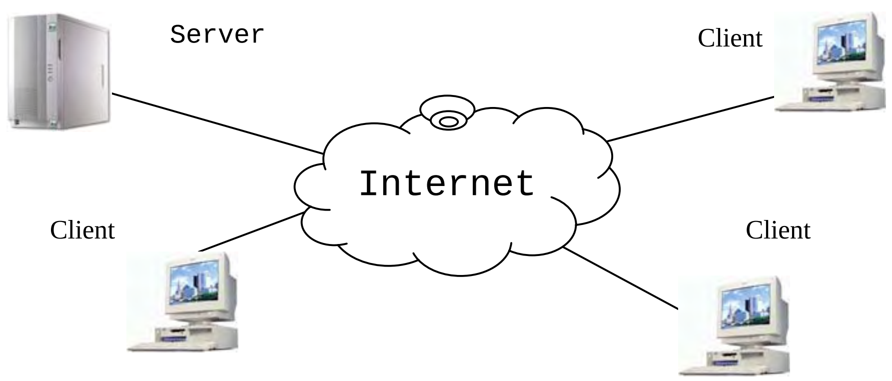


Figure 3.6: *Online gaming system setup*

3.1.4 Video on demand

3.1.4.1 Tests and Means

1.1.1.1.1 Video surveillance

Below in the graphics is show a VoD service deployed on the Unified Platform that is referenced in another task in BROADWAN. It is a Service provided for the end-users which in this case could be used in a surveillance system with remotely connected cameras. There are several connectors where different camera types can be connected and the video streams are manipulated before sent to the surveillance system in a format more suitable for render in the surveillance system application. This kind of service is perfectly suited for dynamic configuration updates which are one of the key-points throughout the BROADWAN applications.

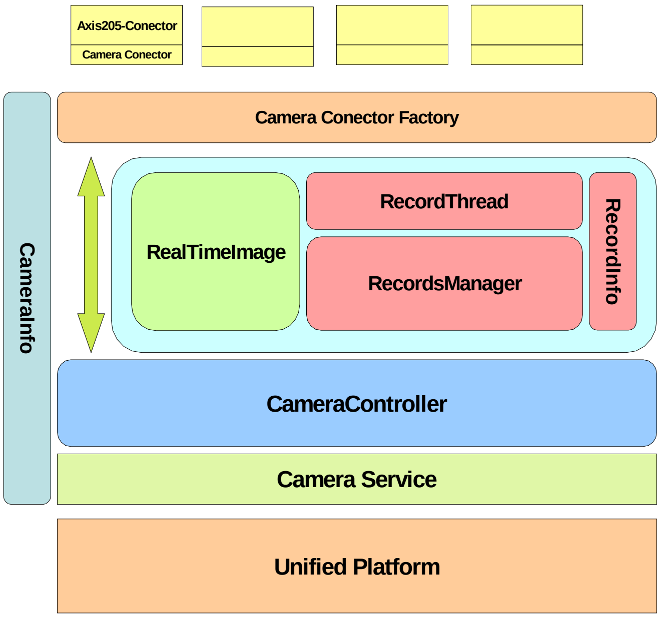


Figure 3.7: *VoD service deployed on the Unified Platform*

The Service
This service is thought to be a suitable application to implement in order to test some important concepts of dynamic configuration of services, more on that later in this section.
The service will be monitoring several cameras and possibly other kind of surveillance equipment which can useful for this kind of services. The service will gather information from these multimedia sources, transform the contents and present the information in a more digestible format in the end-user application.
There are several ways that the end-user application can be set-up. One way is to constantly supervise a certain set of VoD (or Multimedia) sources and another is to dynamically change these depending on a configured scheme. Another possible solution that could be interesting to test depends on indicators that launch different Video streams at demand when defined triggers are activated.
The Platform
The platform that has been described in the Unified Platform section of this document is well suited for a service of this kind. It provides a certain abstraction layer suitable for services of modular nature such as the presented. The platform will be providing additional services for the dynamic network and end-user application feedback. This feedback will then be provided to the platform in order to digest and act on the feedback. The response and actions will be depending on what kind of feedback is provided.
Automatic configuration
Active feedback from the end-user applications, the network, platform or the service will render in new service input which should be used according to configuration of the specific service. This can improve the end-user experience or improve specific network conditions, which is highly desirable and the way to implement the actual feedback mechanism is discussed in the section of the Agent Usage for automatic configuration updates.
The feedback coming in to the service will be treated according to configured rules where the transmission bitrate could be decreased if the network is being heavily congested in order to improve the end-user quality although the it also has decreased. This could be done by lower the resolution, frame rate, or frame size. This is of course a drawback for the end user but not necessarily a bad thing. It is more though as improving the network conditions in favour of other services that might have priority to the network usage.
The other way around is naturally also possible. If there is low usage of the network thus high on available network resources, the surveillance service can treat the feedback indicating the available bandwidth with increasing the throughput for the end-users.

1.1.1.1.2 Multimedia streaming

The Multimedia streaming test environment will consist of a server and one or multiple clients as shown in the following figure.

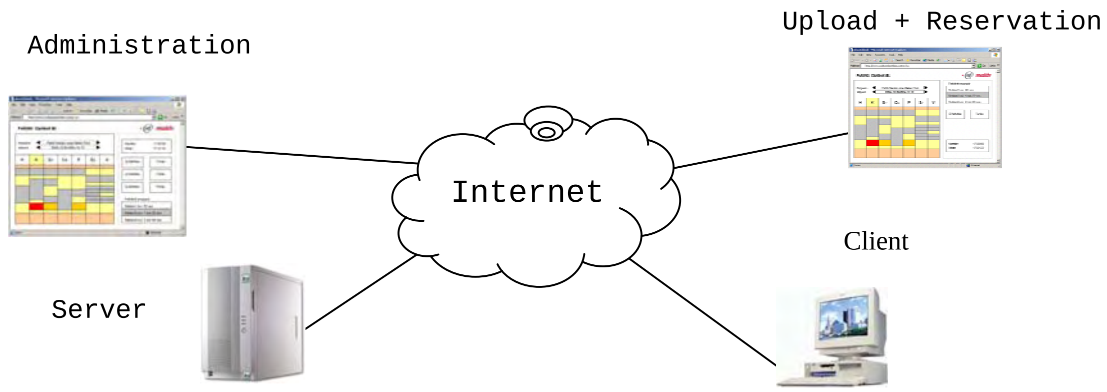


Figure 3.8: *The multimedia streaming test environment*

The software and the media content will by provided by TSR. Upon successful service deployment the clients will receive the broadband video content at the scheduled times.
The service is considered successful with

- good picture and sound quality which relies on low packet loss,
- almost constant availability, which relies on high connectivity.

3.1.5 E-learning

3.1.5.1 Tests and Means

Distance Learning is one of the useful and important services that will be used in a BROADWAN network due to the locality of the end-users. Distance Learning and Distance Workspace are example of areas which are used today although using a completely different setup and normally not in a Hybrid network architecture. One advantage of this kind of service setup is the service quality provisioning, being able to alter the application configuration actively during execution due to network feedback.

System setup

The Distance Learning service can schematically be as the graphics below where we have the end user application connected to the Distance Learning server from where all students are connected. The Network Feedback Service is connected both to the client and the server. Since there usually are several end-users at the same time, the Feedback Service is aware of the identity of all clients in order to provide useful information to the server as well as be able to deliver feedback to the clients.
There will be an interface on both the Server and the clients to permit the communication to the Feedback Service. The implementation of the Network Feedback Service is discussed in a later section in this description.

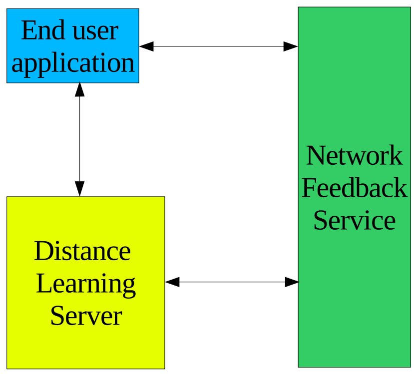


Figure 3.9: *System connections*

Network feedback

The Network Feedback Service will be the plug-in for the Distance Learning Service which is used to collect information from the clients, the server and possibly other network nodes. The Feedback Service can be implemented in many way and we will go for more complete Agent solution that in self is working in a Client-Server fashion where some Agents are specialised on collecting information where others are retrieving this information since their speciality is reading information with this specific contents. The Agent that is working towards the server will provide the Distance Learning Service with information about how the service is performing at any given moment. Then the Service can alter its execution and service for the current situation reflected in the feedback from the Network Feedback Service delivered by an Agent. The dynamic service configuration is further described in the section for the Peer-2-Peer Capacity Demanding service.

Validation

How well can this kind of service be used for this kind of services, such as the given Distance Learning as well as the similar Distance Workspace? This is a question that needs to be answered during a service validation. We need to see if this kind of approach can be applicable to a service of this kind as well as our Feedback approach. The Feedback service that we are planning to use should not only be working for this application in particular, but also in a wider scope for deployment on a platform were we would have several services taking benefit of this feedback service and exchange information between the different application that reside on the same platform. The information that is gathered is not only taking in account the performance of the actual application but could also take in consideration the different applications deployed on the platform. Information can also be collected from other network nodes that not are deployed on the service platform, but still use the different network links and especially the wireless access links which are more bandwidth restricted than fixed networks.

3.1.6 Peer-to-peer networks

3.1.6.1 Tests and Means

The demonstration of the peer-to-peer networks will be based on the works of TSR and MOV. We will user the peer-to peer platform of TSR and incorporate it into the unified platform of MOV. The tests will be carried out by CNRS at Limoges.

The p2p platform offers

- file sharing,
- chat functions,
- displaying of partner availability,
- messaging,
- whiteboard,
- video broadcasting.

These functions will be tested by CNRS personnel.

3.1.6.2 Expected results

As a result we simply expect user satisfaction. This relies on both the performance of the platform and the underlying network architecture. The proper functionality of the features of the p2p platform is heavily based on the network performance, since good quality video broadcast, file downloading and whiteboard usage requires high connectivity, low packet loss broadband infrastructure.

3.2 Demonstration platform 2: Castilla-La Mancha

3.2.1 Platform Description

TCL systems are composed of a space segment and of a ground segment made of a Network Control Centre (NCC), of gateways and of individual Satellite Terminals (STs). Depending on the type of payload in the satellites different network architectures are made possible. The satellite is then connected to another wireless network to reach end users

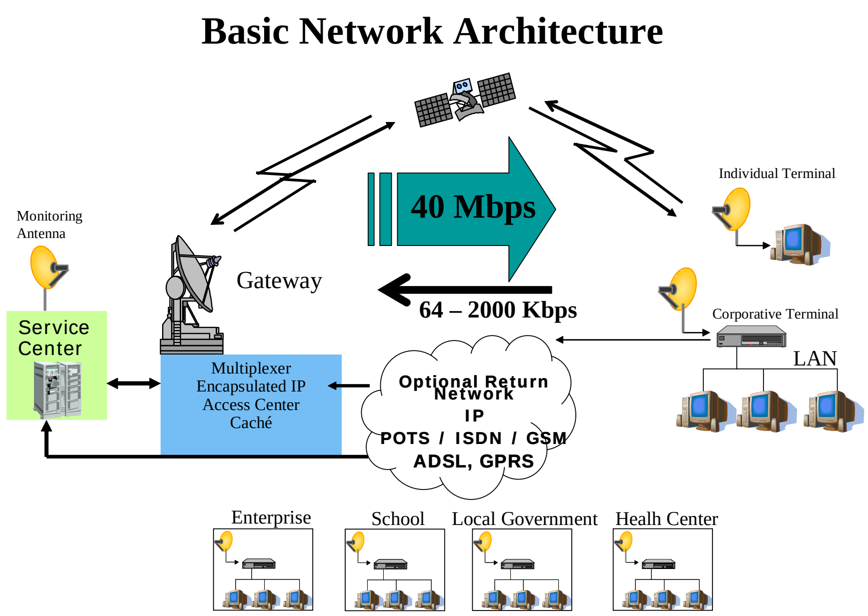


Figure 3.10: *Basic network architecture of DP 2*

Topology used in the Wireless terrestrial network

These networks are, in general, based on a hybrid topology tree-star, which can be distinguished as.

- Transport, connecting the existing network with the distribution points. This part is connected through a point to point link.
- Distribution, connecting the concentration points and distribute the signal to the different points of access through the equipment providing point to multipoint links.
- The third layer allows the connection to the end users, by using the access point and user terminals.

Capacity of the wireless network

The following capacities are assumed in terms of traffic performances assuming the first Fresnel zone is over for 100% of the points:

- Point to point links: 20 Mbps for 10 km.
- Point to Multipoint links: 10 Mbps for 15 km
- Access points: 2 Mbps for 1 km

The system is sized in such a way that allows an easy and fast increase of the capacity up to doubling the initial performances without modifying the aerials.

The link availability is higher than 99.98% for the transport and distribution links and more than 98% for the access links.

Spectrum usage

The range 2.400-2.484 GHz is used preferably for the access links from end user terminals and the range 5.470-5.725 GHz for the distribution and transport links.

The Standard used for the access links is IEEE 802.11. preferably version "a" and, in particular, under the following technical conditions:

- Frequency: 5.470-5.725 GHz (a) 2.446-2.454 (g)

- Modulation: OFDM
- Capacity: 4Mbs for 70 km, 20 Mbps for 10 km (Considering clearance for the 100% of the first Fresnel zone)
- Power at antenna ≥ 20 dBm
- Sensitivity of the receiver (B.E.R. ≤ 10^{-5}) ≥ -92 dBm
- Interface Ethernet 10/100 BT
- Connector RJ45

The Standard used for the point to multipoint links is IEEE 802.11. preferably “a” and, in particular, with the following conditions:

- Frequency: 5.470-5.725 GHz
- OFDM modulation
- Capacity:10 Mbps for 15 km, 2 Mbps for 20 km
  (Considering clearance for the 100% of the first Fresnel zone)
- Power at antenna ≥ 20 dBm
- Sensitivity of the receiver (B.E.R. ≤ 10^{-5}) ≥ -92 dBm

A typical connection at the premises of the user is provided below.

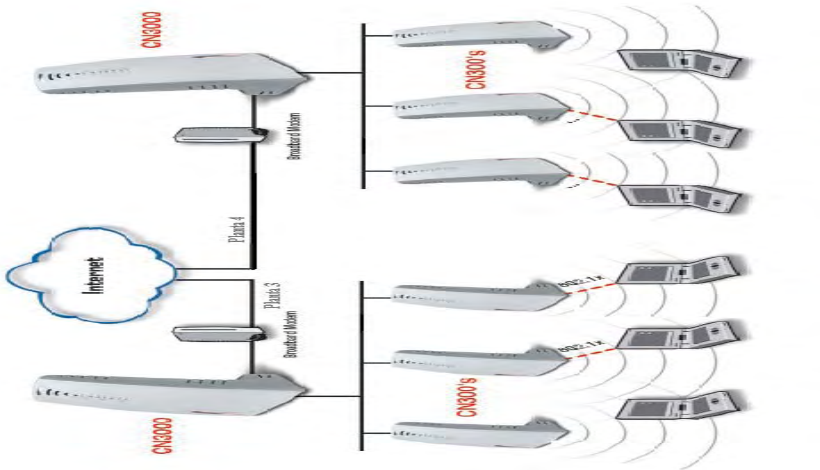


Figure 3.11: *A typical connection at the premises of the user*

3.2.2 E-learning

ING has already started the development of IG-Class over IPv6. It has been analyzed configuration and operation of the low-level streaming technology that the Synchronous IG-Class e-Learning platform uses: Microsoft Windows Media Technologies.

Premises, the IG-Class platform already use this technology, but nevertheless in versions and implementations not supporting IPv6 communication protocol.

3.2.2.1 Tests and Means

First ING will run several tests on Windows Media 9 technology, the platform that fully supports IPv6. Another solution could be Videolan (http://videolan.org).

The system is composed by three main components, each one located in a different computer:

- Window Media Server
- Windows Media Encoder
- Windows Media Player

All of them running last-generation Windows Operation System, IPv6 protocol has been set up manually, in addition to the previous named applications, since none of them were installed by default.

System configuration

A complete Windows Media System is formed by three independent software components that can be executed in one or several computers. By now, each component is executed in a different computer.

Windows Media 9 Server: this component takes the multimedia sequence from the encoder and serves it to the players. It is an optional service incorporated in Windows Server 2003, not available for another operating system. It has been chosen Enterprise Edition, as it is the unique one that support sequence transmission using Multicast IP protocol, as well as Unicast IP. From now on, we will call this the *Server* computer.
Windows Media 9 Encoder: this component is in charge of creating the multimedia sequence from the system devices. This sequence will by transmitted to the server computer by using the HTTP protocol over IPv6. The operating system running on computer is Microsoft Windows XP SP1, and from now on, it will be called the *Encoder* station.
Windows Media 9 Player: It is in charge of playing the multimedia sequences, it uses MMS as the transport protocol. The operating system of this computer is also Windows XP SP1, and from now on, it will be called the *Player* station.

Network configuration

The Server computer has two network adapters installed, whereas the Encoder and the Player only have one. After installing the operating system, we installed IPv6 protocol in each computer, following the instructions provided by Microsoft. Then, the TCP/IP (IPv4) protocol has been deactivated from each network connection, in order to make sure that all communications between the computers are made over IPv6 protocol and no IPv4 traffic exists.
IPv6 network addresses has not been manually specified, since the protocol automatically assigns local addresses to each computer, based on MAC Address of each network adapter.
The following connection topology between the computers was been set up, in order to isolate the Encoder from the Player. The following image shows IPv6 address of each interface, and the protocols used to transport the multimedia sequences.

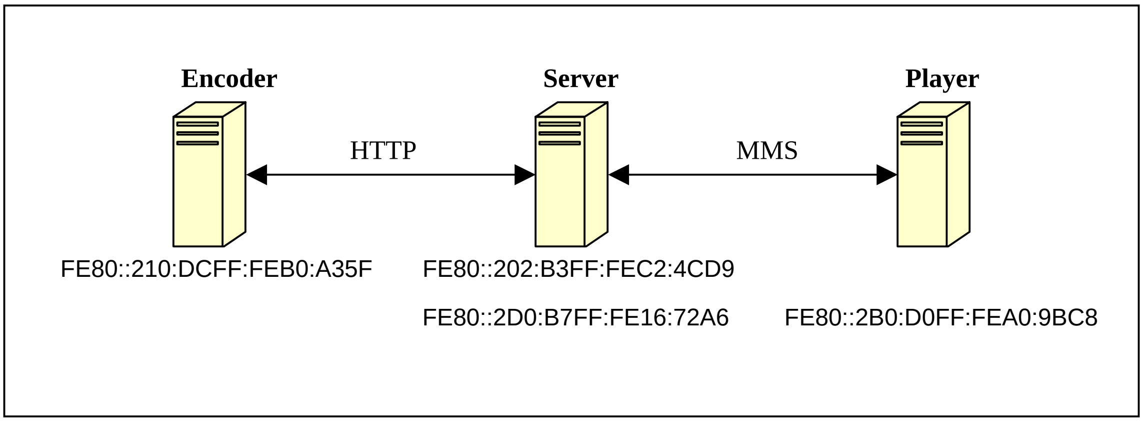


Figure 3.12: *System and Network Architecture*

Multimedia configuration

In order to capture and to encode the multimedia sequence we have installed a USB Logitech Quickcam camera and we also make use of the motherboard integrated sound adapter, verifying that both work properly.

In the Player station a Creative SoundBlaster Live sound card was installed, to play the audio track from the multimedia sequence. The server has no multimedia adapter installed, as it is not needed for the operation of the Windows Media Services.

Test accomplishment

Tests done to verify the performance of Windows Media 9 over IPv6 were the following:

- Generation of a multimedia sequence in the Encoder.
- Transmission of that sequence to the Server using HTTP (Unicast) protocol.
- Retransmission of the sequence using MMS (Multicast) protocol.
- Player connection to Multicast channel to receive the sequence.

In order to reduce the waiting time we changed the player network buffers and leave them in 1 or 2 seconds. The optimal value will be the one that allows receiving the multimedia sequence without cuts.

In order to verify the system stability, it was left running under the previous conditions for 24 hours without any reception or playing problems detected.

The Demonstration Platform 2 will be based on the following architecture:

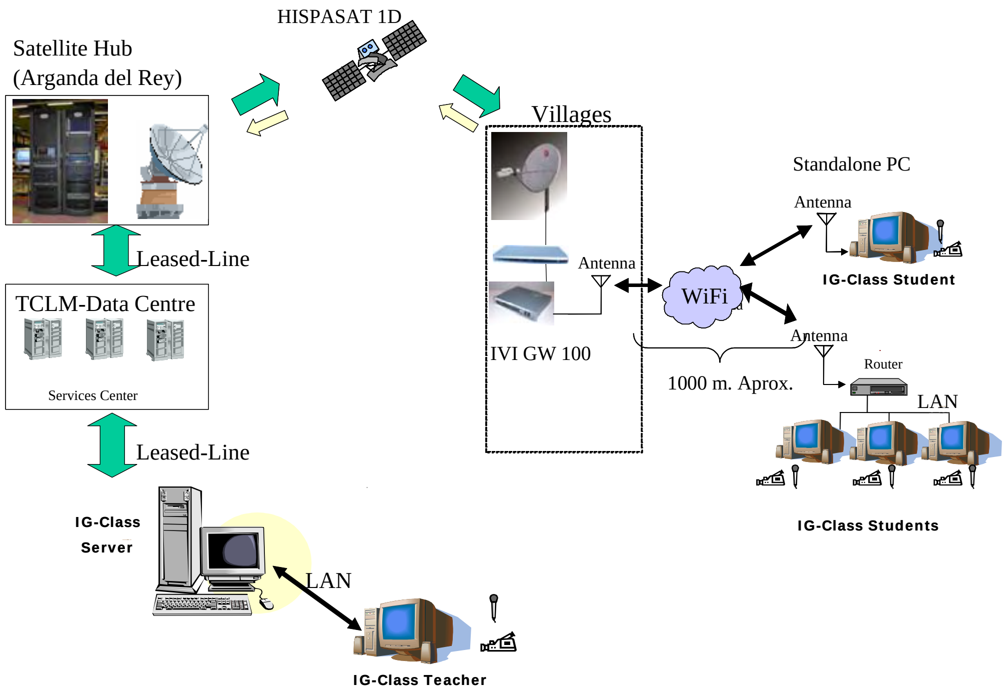


Figure 3.13: *Demonstration Platform 2 architecture*

On this architecture ING and TCL will test new IG-Class version features to minimize coding delay in order to define the best parameters for Teacher's codification, Server Streaming and Students decoding and coding.

Once this system architecture is stable through the two-way satellite platform ING will add students connected to the system through terrestrial accesses to compare performance and interactivity.

It will also be tested teacher's contribution through a two-way satellite terminal to test uplink demanding services. The teacher's workstation produces a 512kbps average streaming.

3.2.2.2 Expected results

It is expectable the codification and streaming delay in the actual IG-Class version will be dramatically reduced.

Especially interesting for the platform interoperability is the interaction between students accessing the system with different technologies, bandwidth and delays.
About teacher contribution through a two-way satellite terminal one cannot expect to have a good performance because the bandwidth is shared among many terminals (1:50, 1:100, 1:200) in the uplink, and is really expensive to use a dedicated bandwidth. Anyway it will be tested using several contention rates.

3.2.3 E-learning multicast

ING and TCL has a long experience dealing with e-learning using multicast through satellite platforms, but we have never tested it through two-way satellite terminals which architecture is much more complex, at least at satellite HUB premises.

3.2.3.1 Tests and Means

Following the same scheme used in the previous test multicast content delivery will be also tested. ING and TCL will work together to configure the two-way satellite in order to allow multicast traffic to "flow" through it.

3.2.3.2 Expected results

We hope we can make it work based on our know-how on satellite platform and the manufacture (Nera, ViaSat) commitment to aim us in this task.

3.3 Demonstration platform 3: Oslo

A technical demonstration of multicast is planned using the 42 GHz test platform located in Oslo. The test-platform is currently implemented using the DVB-RCS standard, but for the purpose of these tests this is not regarded as a limiting factor. The rationale behind the demonstration is to test and verify the operation and functionality of multicast techniques using an IPv6 system. For practical reasons, the initial tests will have to use IPv4, but will later be migrated to IPv6.
The application selected for these tests is first the distribution of digital TV using multicast techniques. The performance of such a system will be evaluated against a system using the simpler unicast technique.
Later on, the tests will be extended to use layered multicast techniques.

3.3.1 Platform Description

In Figure 3.14, an overview of the system is shown. The system consists of a Base-Station (BS), and one or more User-Terminals (UT) connected to the BS via the air-interface.

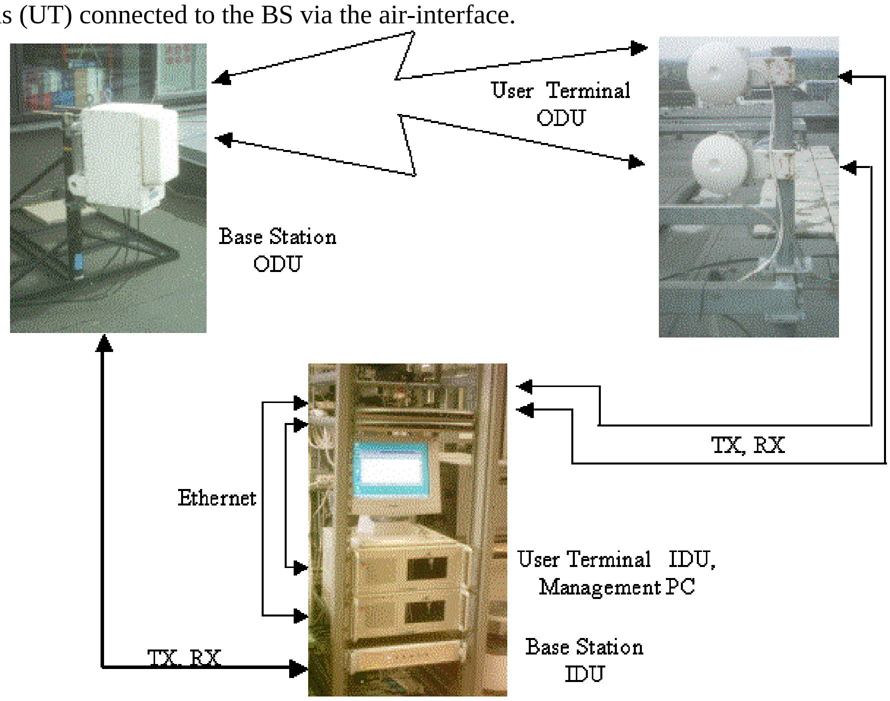


Figure 3.14: *Experimental set-up for the multicast test*

3.3.2 Base-Station

The BS consists of a Forward Link Subsystem (FLS), a Return Link Subsystem (RLS), a Reference and Synchronisation Subsystem (REFS), the Radio-Frequency and Antenna Subsystem (RF) and a Network Control Centre/Network Management system (NCC/NMS). This is detailed in the block-diagram of Figure 3.15.

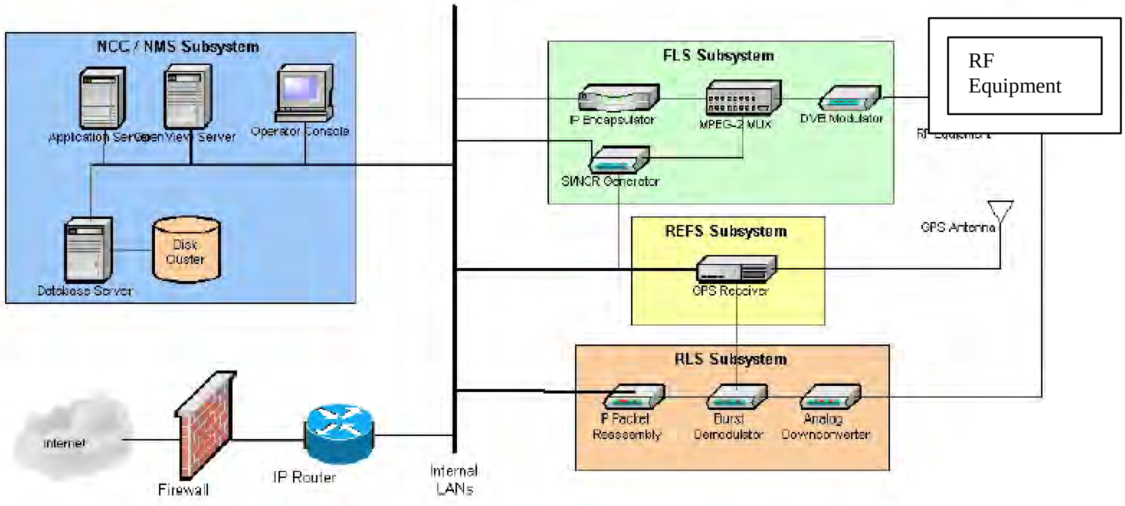


Figure 3.15: *Base-Station, Block-diagram*

The BS is connected to the terrestrial network through a firewall and a standard IP-router. The incoming IP packets received at this interface are encapsulated into MPEG-2 packets according to the DVB-S standard. This encapsulation is done in commercially available equipment designed for this purpose. System broadcast messages and terminal-specific signalling from the NCC is multiplexed onto the forward transport stream, which is transferred to a standard DVB-S modulator and sent to the RF sub-system for translation to the appropriate frequency (channel) in the 42GHz band.

The BS RLS functions are basically the reverse of the FLS functions, where the received signal is translated from RF via an IF (950-1450 MHz) to base-band and demodulated. The return channel signalling information is split, and the end-user IP-packets are restored and transported to the terrestrial network interface.

The REFS delivers the synchronisation and timing information for the BS for synchronisation of the network. The system timing is derived from a GPS receiver located at the BS.

The RF subsystem translates the signal from the DVB-S modulator at IF to the appropriate frequency (channel) at RF (42 GHz) for the transmit direction, and from RF-to-IF for the receive direction.

The NCC/NMS provides the interface for operator network management, the radio resource management, traffic control handling and generation and distribution of DVB-RCS system tables.

3.3.3 User-Terminal

The User-Terminals (UT) consists of two main units, an out-door unit (ODU) and an in-door unit (IDU). These are connected via a coaxial cable.

The ODU consists of an IF-to-RF and an RF-to-IF conversion circuitry for the transmit and receive directions respectively, and the power-amplifier and the low-noise amplifier associated with the transmission and reception of the signals. In addition the antenna-element is integrated.

The IDU contains the DVB-RCS modem, which also include the interface to the local network.

3.3.4 Multicast demonstrations

3.3.4.1 Tests and means

Streaming of TV/Video

Distribution of digital TV signals using multicast techniques are an emerging application that is of interest in the BROADWAN system. Unlike the current TV distribution systems, which are either based on terrestrial transmission systems (cable or over the air) or satellite, using broadcast techniques (all channels are broadcast simultaneously), the bandwidth available in a BFWA system will severely limit the total number of channels transmitted. Today, a standard definition TV signal is transmitted using roughly 4 – 6 Mbps, which will increase to 20 Mbps per channel as soon as high definition television is demanded. This will quickly fill the capacity of a sector in a BFWA system.

Using IP multicast techniques for the distribution of television could be a solution to this problem, since the programme material is only sent once over the radio-hop, and the users interested in this material will subscribe to this part of the transmission.

In Figure 3.16, a more detailed illustration of the network architecture that will be used for multicast tests is given. The IP router, the management PC and the associated DVB modulator and RF circuitry are located at the Base-Station side. The out-door and in-door units are placed at the User-Terminal side, with a set-top box and a TV-set. A standard PC is used for the monitoring of IP-traffic.

The various TV-channels are set up as multicast streams from the head-end when a subscriber/multicast client asks for a channel. The actual set-up of the stream is done by sending an IGMP (Internet Group Management Protocol) message to the multicast source and after the terminal has been authenticated and authorised to receive the multicast data is sent.

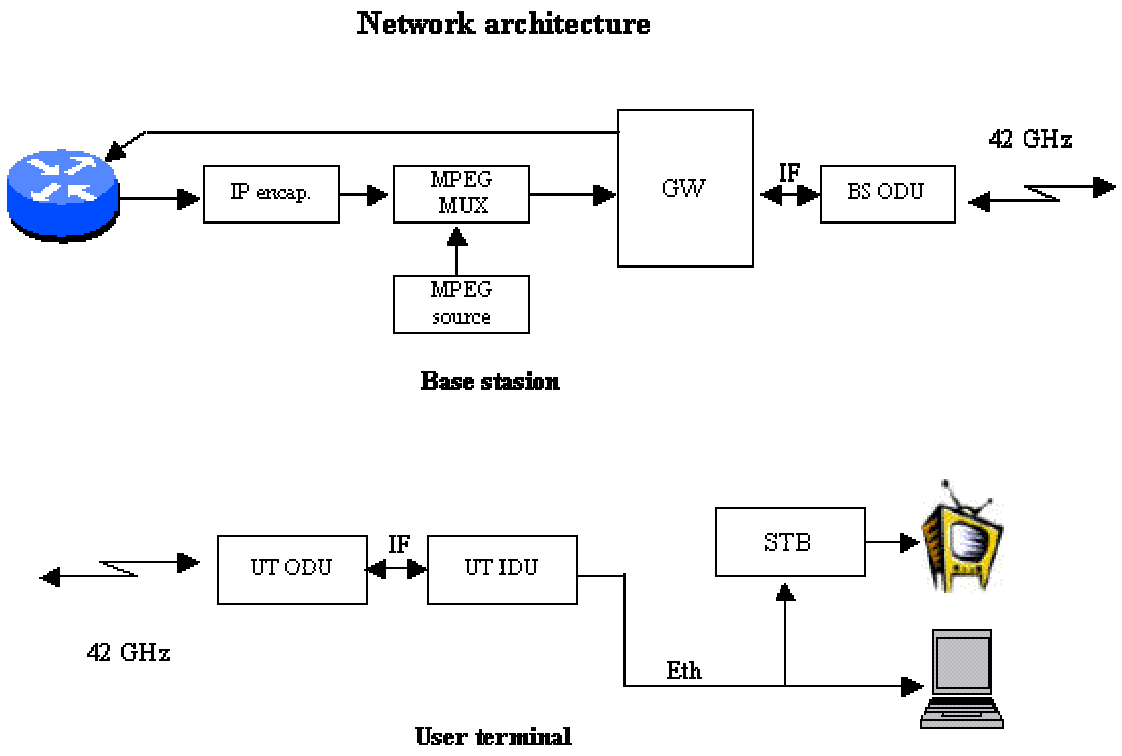


Figure 3.16: *The network architecture for multicast streaming experiment*

Layered multicast

In the future, the distribution of real-time audio and video is expected to be performed using layered multicast techniques. This method will allow the quality of the programme material to be dynamically adjusted to the link conditions and available processing capabilities in the user-terminal. The different layers carries material with different levels of quality, and makes it possible for equipment employing adaptive coding and modulation techniques to transmit the level of quality appropriate for the current link conditions.

Layered video transmissions require hierarchical video/audio coding schemes that scales to a large number of users, and which supports different quality levels without any noticeable bandwidth overhead (e.g. MPEG-4). It will also allow every user to change the quality of the video/audio stream at any time, in order to either have the best possible quality or to save bandwidth and, presumably, costs [1].

3.3.5 Performance

The main motivation for using multicast techniques in a network is the efficient utilisation of available resources, i.e. bandwidth. Multicast will reduces the overall bandwidth needed for the transmission of programme material since packet duplication only occur when paths to multiple receivers diverge, given that many end users receive identical data traffic.

On the other hand, the use of multicast carries some penalties. Control traffic in the network used to update necessary multicast soft state, such as memory usage due to state in the network elements, and possibly increased CPU usage due to multicast processing, will have a negative effect on the total efficiency. In addition comes the complexity of deployment and management of multicast groups and multicast trees etc.
The bandwidth savings will depend on the number and distribution of receivers sharing the network resources, and on the shape and topology of the multicast distribution trees.
It is found [2], that bandwidth savings per multicast group could be significant even for small multicast groups. For group sizes as small as 20 – 40 receivers, a reduction in number of links traversed is found to be about 55 – 70 % compared to separately delivered unicast streams. For groups of about 150 users, the predicted saving is 75 %, and this can get as high as 90 % when the number of receivers is more than 1000.
Obviously, the overall bandwidth efficiency that may be achieved by using the multicast technique in the network will depend on the scenarios that are foreseen. A study of the performance using the multicast technique versus the unicast technique for a given traffic load will be carried out. The results will be compared to other studies of the efficiency of the multicast technique.

3.3.6 Expected outputs

The output of these multicast experiments is expected to give a verification of the multicast technique and the feasibility of using such techniques in a BFWA system in the 40 GHz band will be conspicuous. The tests will lead to an increased knowledge about multicast service behaviour, and on how the layered multicast techniques will work.
In addition, information on the system performance with regard to bandwidth utilization will be obtained. It is expected that multicast will reduce the overall bandwidth demand when several users requires the same information, compared to unicast, and hence an increased efficiency will be achieved.

3.4 Test-beds: Buckingham and Salzburg

3.4.1 Mobile ad-hoc test-bed (UBU)

3.4.1.1 Purpose of a hybrid IPv6 wireless network test-bed

BROADWAN has brought many new techniques and challenges in terms of network planning, management, services and applications for all-IPv6 heterogeneous networks. It is absolutely essential to demonstrate and verify these new techniques for the interests of the project and future development. The test-bed in the University of Buckingham is a part of this demonstration test-bed. It is aimed to establish an all-IPv6 infrastructural, wireless Ad-Hoc network incorporating the provision of dynamic reconfiguration. The dynamic reconfiguration of individual elements of a heterogeneous network and QoS guarantees will be key features. While serving as a showcase for many general applications and how seamless transition from IPv4 and nomadic user roaming can be achieved, the primary objective of this test-bed is to demonstrate some of the novel IPv6-based services and applications derived from the BROADWAN. It is not directly concerned with user migration to Ipv6.

3.4.1.2 Architecture of the test-bed

Based on the above purpose and consideration, an infrastructural wireless mobile ad-hoc network test-bed is set up at the University of Buckingham. Its architecture is shown in the following figure.

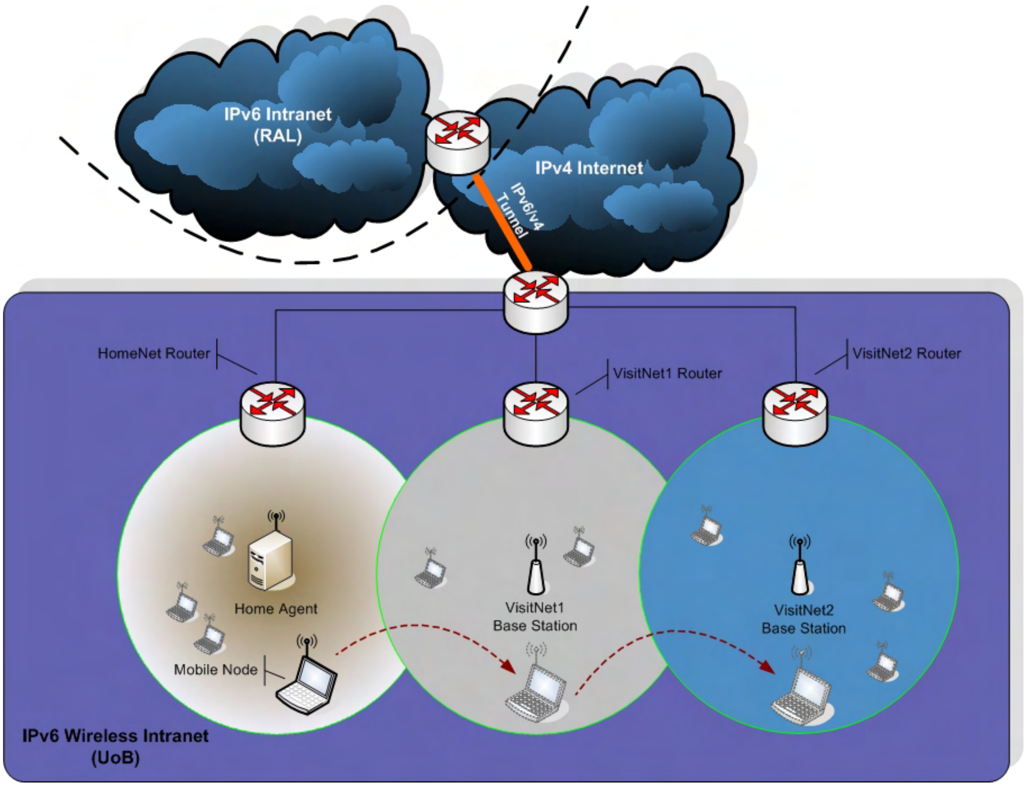


Figure 3.17: *The Architecture of Heterogeneous Mobile Ad-hoc Test-bed*

The test-bed at the University of Buckingham consists of three access points, several routers and mobile nodes. The access points functions as a base station in a local access site. All routers are IPv6-enabled Linux machine with kernel 2.4/2.6 and wireless network cards. The wireless interfaces are IEEE 802.11a/b/g compatible and work at both 2.4GHz and 5 GHz. The IPv6/v4 tunnel is ready for connection with the trial systems at the Rutherford Appleton Laboratory via the public Internet. AODV routing protocol and Mobile IPv6 protocol are currently running on the test-bed. The Router Advertisement protocol is used to reconfigure nomadic nodes in the wireless access areas. All nodes adopt the dual IP stacks strategy to get access to the Internet.

3.4.1.3 Technical consideration and requirements

The major concerns for setting up such a heterogeneous wireless ad-hoc network test-bed lies on the issues of IPv6 getting access to the Internet, ad-hoc routing strategy, network reconfiguration and verifying new services.

IPv6 and access to the public Internet

It is no doubt that the test-bed is an all-IPv6 network with public Internet access. It is unfortunately that the Internet is still dominated by IPv4 networks. There exist several options for IPv6 networks internetworking with the IPv4 Internet and other IPv6 networks. Dual-stack and IPv6/v4 tunnelling are the most commonly used solutions. A dual-stack node is flexible to connect to both IPv6 networks and IPv4 networks. While nodes within IPv6-only networks keeping lightweight operation systems, they can access other IPv6 networks via the Internet with IPv6-over-IPv4 tunnelling technique. However the other end of the communication has to be aware of the existence of the tunnel and an equivalent tunnel point has to be in existence. For reasons of making connection with the Internet flexible and easy access to other IPv6 networks via the Internet, it is determined that the test-bed makes use of both techniques. It is also a compromise especially when many services and applications still rely on the IPv4 stack, although the aim of the test-bed is to verify novel applications for IPv6 derived from the BROADWAN project.

Ad-Hoc routing strategy

When a node roams from one site to another, its connectivity and identity at the network layer need to be kept unchanging. The Mobile IP standards are specifically designed to resolve these problems. For the test of nomadic user roaming, the test-bed deploys the latest Mobile IPv6 implementation for Linux. The implementation is based on the IETF standard “Mobility Support in IPv6” (RFC 3775).

A dynamic ad-hoc network is formed when nodes in a certain site and/or area closing to each other and can roam, join and leave the network freely. In this case, there is more likely a route existed for node pairs to communicate with. The need of a base station intervening is less likely. This is judged as an advantage of the heterogeneous wireless ad-hoc network with the support of infrastructure. Currently there are several different ad-hoc routing protocols for choosing. In many aspects the AODV (ad-hoc on-demand distance vector) routing protocol is considered as performing better than other protocols. The test-bed is therefore deployed with an implementation of the AODV routing protocol developed at the University of Buckingham.

Multicast and streaming applications
The test-bed will use the current Linux IPv6 implementation for multicast services. There are plenty of streaming applications based on the Linux IPv6 multicast available for the evaluation of the test-bed functionality.

QoS issue
It is a sufficiently difficult challenge to provide a dynamic mobile ad-hoc network with QoS guarantees. Such guarantees, if there is any, may be very difficult specifically when considering how dynamic this network is. It requires acceptable channel conditions, QoS-aware mechanisms for channel access, measures for congestion prevention as well as relatively stable and reachable nodes, etc. Most of them rely on a smart MAC layer and interconnection with the network layer. The test-bed utilises the normal QoS components available to the Linux platform at the network layer. Any MAC layer QoS provision is worth testing.

3.4.2 All-IPv6 test-bed (USA)

A test-bed has been implemented on USA premises, which is intended to evaluate novel services to be used within the BROADWAN project, based on IPv6 protocol. This test-bed is not to be understood as a demonstrator, but mainly to prove the idea behind the concepts to be verified and the readiness in terms of implementation status according to existing standards.
According to the network architecture planned in the BROADWAN project, the USA test-bed is intended to simulate this environment using a set of PCs emulating different nodes of the network. Different PCs are used for dedicated simulation of network components such as a router or a physical link. Simulation of physical (wireless) link behaviour is highly useful in case of testing the behaviour of (mainly wireless) links in conditions of high traffic load on the air interface or bad signal quality or even high propagation delays (in case of DVB-S links).
As shown in the logical test-bed illustration below, a network consisting of a backbone with two different LANs (L1 and L2) is connected to a backbone router (R3). This router is connected via a (fixed or wireless) link simulator to two different user connecting routers (R4 and R5), serving different subnets. As an option, the link simulator may be replaced by a real wireless unidirectional link like DVB-S or DVB-T.
Most components (except LBL simulator) are connected via Gigabit Ethernet (connections labelled “C*” in the test-bed illustration) to provide maximum physical link performance.
Following scenarios can be realised in the test-bed:

- S1 – single limited bandwidth link, utilising connections C1 and C4
- S2 – point-to-multipoint limited bandwidth link, utilising connections C1, C4, and C5
- S3 – two limited bandwidth links, using connections C1, C2, C4, and C5
- S4 – single limited bandwidth link with intermediate feed forward router, using connections C1, C5, and C6
- S5 – an additional unidirectional link (DVB-S or DVB-T), using connection C3

This list represents a subset of possible configurations, identified to be most commonly used and of high significance with respect to the „real world”-implementation of the BROADWAN network.

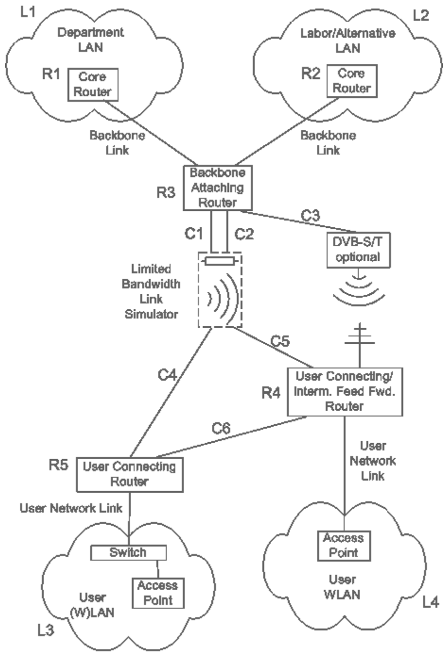


Figure 3.18: *Logical layout of USA test-bed*

Hardware layout of the USA test-bed, showing PCs numbered 3-8 corresponding to R1, R2, R3, LBL simulator, R4, and R5 in the logical layout. The second picture shows the cabling of the machines, especially the LBL simulator having 4 Ethernet connections (3^{rd} in row). Not included in the picture is the DVB-S / DVB-T option (linked on C3).

Figure 3.19: *Test-bed equipment*

3.4.3 Collaboration points

Collaboration points have been identified between USA test-bed and demonstrators operated by Telenor/Nera and Limoges demonstrator platform. It is planned, upon successful validation of services in the USA test-bed, to install IMPPS multicast Web caching service on the Limoges demonstrator, and the adaptive video multicast service on the Telenor/Nera demonstrator, in order to give these novel multicast services public visibility.

3.4.4 IMPPS multicast web cache/proxy

Tests will be performed on logical parts of the multicast Web-caching and proxy services, evaluating the performance in the test-bed network and issues related to IPv6. These parts are

- Multicast peer caching: all peers authenticated to the network will receive cache updates from other peers, reducing the probability of retransmission for different peers while optimising link bandwidth usage and increasing cache hit ratio.
- Filtering: implementation and testing of automatic filter rules.
- Cache Preloading: Speculative installation of data in a cache utilizing layered multicast transfers, in the anticipation that it will be needed in the future.
- Prefetching: Cache-initiated speculative installation of data in a cache in the anticipation that it will be needed in the future.
- Feature tests
    - Encryption
    - Reliable Multicast
    - Layered Multicast
    - Audio/Video Stream-Splitting
    - Transparent Content Compression

IMPPS Expected results

- Validation of functionality in an all-IPv6 network
- Performance validation of
    - Broadcast link efficiency
    - Prefetching efficiency
    - Preloading efficiency
    - Cache efficiency.
- Efficiency of content filtering

3.4.5 Adaptive video multicast

Within the test-bed, following test are planned to be performed:
Layered Multicast of MPEG streams:

- Partitioning of the MPEG stream into layers, by separating the video into a meta-data layer, base and enhanced video layers, and audio layers. This is achieved by
    - Separating the video into I-Frames, P-Frames, B-Frames, resulting in one base layer and three video layers.
    - “Scalable encoding”, for example spreading video pixels, frames, DCT coefficients over several layers, or using error correction properties of the encoding format (for MJPEG2000).
- Transmission over layered multicast channels.
- Investigation of transmission parameters and behaviour in various network conditions.

General-purpose file transfer over layered multicast:

- Find or develop a suitable implementation, focusing on
    - Simple usage
    - “Point-and-click” application for technically "marginally-involved" users.
- Should support layered multicast, cope with multicast problems.
- Evaluate usage of one-time-transfer vs. carousel mode.

Adaptive video multicast expected results

- Prove-of-concept for adaptive video multicast
- Conclusion on performance of tested coding/layering techniques
    - Bandwidth usage
    - Reception quality
    - Required calculation (processor) performance
- End-to-end verification using different network quality parameters

3.4.6 Service discovery

- Find a viable combination of the standardised service discovery protocols to be tested
    - combine mainly the Internet and digital television service discovery world.

- Identify custom extensions that might be necessary to existing protocols to fit the BROADWAN project requirements
- Assembly and distribution of meta-data information
- Reception and analysis / receiver behaviour
  - Auto-join of discovered service
  - List view of available services

Service discovery expected results

- Definition and implementation of service discovery mechanism
  - Service discovery meta-data definition
  - Service discovery protocol
  - Service discovery transport
- Find means to assemble / define meta-data
- Definition of behaviour of reception devices / user receiving service discovery information

3.4.7 IPv6 auto-configuration

Auto-configuration in IPv6 is the process of querying for and setting of basic stateless network parameters when a node attaches to an existing and configured network. This is in contrast to ad-hoc-configuration where a group of hosts spontaneously form an isolated network.
From the lessons learned form IPv4, auto-configuration is one of the design criteria for IPv6. The stateless auto-configuration procedure for IPv6 nodes was part of the IPv6 RFC right from the beginning and is very successful in configuring hosts in IPv6 networks. While also a statefull address auto-configuration mode was designed later on it is seldom used and generally considered unnecessary.
What works well for host configurations had no counterpart at all on network/router level where whole networks should be automatically distributed to attached routers. Only now effort has been put onto this topic resulting in the idea of prefix delegations through the DHCPv6 protocol. Unfortunately current RFCs deal only with one hierarchical level, that is, delegating and delegated routers are directly connected.
In a network infrastructure as complex as the BROADWAN network some hierarchical levels are expected where on the other hand it is quite reasonable to propose that the structure can be presented in a graph without cross connections (network is not meshed). For such a network it seams quite suggestive to extend the single hop prefix delegation into a hierarchical multi-hop prefix delegation.
Most of the IPv6 concepts are still very new, not deeply debugged and the implications are not well enough understood. Therefore, it is necessary to understand different implementations and their limitations. The focus will be on the freely available operating systems Linux and FreeBSD. This is namely because they are on the bleeding edge of technology and they offer the opportunity to modify and adapt to different requirements.
The support for prefix delegation and automatic router configuration possibilities must be checked and extended where necessary. Although it is expected that a working prefix delegation implementation for single hop environments can be found, it is clear that none will exist for multi-hop environments. Therefore, some simulation and/or additional coding must be performed in order to verify the suitability of the proposed concepts.

IPv6 auto-configuration tests

- Identification of full-IPv6 and, if possible, IPv6-only supporting OS.
- Identification of software capable for prefix delegation
  - in single hop environments
  - and in multi-hop (hierarchical) environment.
- Simulation and/or implementation of prefix delegation algorithms.
- Impact of hierarchical prefix delegation on mobile IPv6 (MIPv6).

IPv6 auto-configuration expected results

- IPv6 standards are still emerging so complete implementation are not available. IPv6 is always implemented as an ad-on to IPv4 thus no broadly available IPv6-only implementation exists. Different limitations on different architectures: BSD is feature rich but Linux comes with more sophisticated designs and algorithms.
- No full featured implementation is available. Most features are available in the KAME software suite for BSD derivates.
  - Single hop prefix delegation works well but routers need fully static configuration.
  - Multi-hop prefix delegation does not work without additional coding. E.g. prefix delegation is handled by DHCPv6 but the KAME implementation of DHCPv6 server and clients don't

interact when running on the same machine. No interface to the router advertisement daemon exist either.

- Case studies should show that the idea of hierarchical prefix delegation is reasonable. Further more corner cases and problematic configurations should be identified.
- On networks already configured to support MIPv6 minimal impact is expected for mobile hosts. Access routers that are mobile by itself are a major problem because there requirements do not fit into the prefix delegation framework. On the other hand cases of mobile access routers are not identified up to now so this area may not be addressed in the BROADWAN project.

3.4.8 Mobility

In the "*Mobile ad-hoc test-bed*" described in Section 3.4.1, nodes can roam freely in two possible modes, the intra-site mode and inter-site roam mode. Therefore we use two ways to tackle the node mobility issue. For the inter-site roam the Mobile IP protocol is used, and for the intra-site roam the node can either keep in using Mobile IP or working in ad-hoc mode. To make terms less ambiguous, here mobility means the inter-site mobility. The mobility of ad-hoc nodes will be described in Section 3.4.9. Mobile IP provides inter-site roaming mobile nodes with mobility at the network layer (L3). Combined with dynamic reconfiguration, mobile nodes can freely move from one site to another without worrying about the configuration of mobile nodes and the loss of connection if a Layer-2 transparency can be achieved. This is certainly true for our test-bed.

3.4.8.1 Test and means

So far the test-bed has been used to verify and demonstrate the nomadic user roaming and transition from IPv4 to IPv6. Experiments including performance assessment of infrastructural wireless ad-hoc network, dynamic reconfiguration of mobile nodes, multimedia streaming applications and QoS provision are all in the scope of further investigation of the test-bed. The suggested tests and means are explained in follow:

- Performance assessment of inter-site node mobility

  Effort will be concentrated on the assessment of node mobility when nodes move between different sites and the correspondent nodes are located in different domain (e.g. within the test-bed or in the public Internet). The assessment is based on different node movement models and the criteria of assessment are those tightly related to performance of mobile network such as:
  - Hard hand-off performance
  - Packet delivery performance

- Dynamic reconfiguration of mobile nodes

  The major concern for dynamic reconfiguration of mobile node in the test-bed is how the node's identification (IPv6 address) is allocated and configured when the node roams from one place to another. A layered global unique local IPv6 address architecture is used in the test-bed. While the uniqueness can be reasonably and mathematically assured, the accessibility of the global unique local address is to be tested thoroughly via communications amongst intra-local network, inter-local networks and the public Internet.

- Multimedia streaming applications

  Multimedia streaming applications are used to test usability of the mobile ad-hoc networks and interoperability of heterogeneous networks when the mobile user is in constant roaming within the geographic coverage of network service. The planed streaming applications include:
  - Streaming audio
  - Videoconference or other suitable video streaming applications.

- QoS provision and improvement

  The most severe QoS problem in the scenario of Mobile IP application is that the mobile node can only hear from one base-station during the transition from one service site to another (assuming the mobile node is working in associated mode). This is the so-called hard hand-off problem. A longer hard hand-off delay will cause temporary connection loss and service interruption. Although it may be acceptable for time insensitive service like file transfer and email, it certainly has severe effect on the time-critic application such as multimedia streaming. Therefore the test for QoS will be mainly focused on:
  - Evaluation of the hand-off delay and its effect on time-critic applications,
  - Improvement of QoS by means of an improved client hand-off decision making mechanism.

There are no restrictions for other novel applications derived from the project to be tested on the mobile ad-hoc test-bed as long as they are compatible with IPv6.

3.4.8.2 Expected results of mobility service

- An evaluation of node mobility performance in the test-bed
- Definition and implementation of a mobile node dynamic reconfiguration for infrastructural mobile ad-hoc network
- Definition and implementation of QoS improvement for hard hand-off mobile nodes

3.4.9 Ad-hoc networks

In the "*Mobile ad-hoc test-bed*" described in Section 3.4.1, mobile nodes can communicate with each other by either associated with the base-station or self-organised ad-hoc network. When a mobile node needs to contact with other nodes in a different area or in the external Internet, the infrastructure including the base-station is used. Note not all mobile nodes require connections other than connections with nodes in the same area. In this case, the ad-hoc network plays a vital role to provide with inter-connections for mobile nodes in the same area. Ad-hoc networks could establish node connections by different routing protocols. AODV is such a protocol that is widely used in the ad-hoc networks. When a source node needs a route to a destined node, it sends a route request (RReq). If the destined node or some other nodes knew a fresh route to the destined node hear the request, they will reply the request with a route reply (RRep) and thereafter a node-to-node route is set up. Because the route may pass several mobile nodes (i.e. it is a multi-hop route) and the nodes are in a dynamic roaming state, the route could be expected only last a short period as well as unreliable. The more hops the route passes, the weaker the route is. Our routing strategy in the described mobile ad-hoc network test-bed takes a step further. One major feature of our test-bed is the use of backbone to assist the AODV routing protocol to establish multi-hop routes and deliver packets to remote nodes that are not reachable with normal AODV protocol. With this feature, we can expect not only longer multi-hop routes but also higher success route ratio and improved packet delivery ratio compared with the widely used normal ad-hoc routing protocol.

3.4.9.1 Test and means

- Performance assessment of infrastructural wireless ad-hoc network

  Effort will be concentrated on comparison of the infrastructural wireless ad-hoc test-bed's performance with that of no-infrastructure supported wireless ad-hoc networks. The comparison is based on different node movement models and the criteria of comparison are those tightly related to the ad-hoc routing protocol such as:
  - Average length of routes
  - Distribution of the shortest available path length
  - Packet delivery ratio
  - Ratio of reachable nodes
  - Routing traffic overheads.

  Instead populating the test-bed with many mobile nodes, the JiST/SWANS (Java in Simulation Time / Scalable Wireless Ad-hoc Network Simulator) simulation platform with an extension of backbone support is used to simulate different network scenarios for the performance comparison. A preliminary simulation has given us very encouraging result of the infrastructural mobile ad-hoc network.

- Nodes connectivity test with an AODV routing protocol implementation for IPv6 on Linux platform developed at the University of Buckingham. It is also planed to test the node connectivity with multimedia applications such as streaming audio and video.

- Dynamic reconfiguration test of the mobile nodes in ad-hoc network. The objective of this test is to examine the auto-configuration scheme suitable for both Mobile IP and ad-hoc network.

3.4.9.2 Expected results of Ad-Hoc networks

- A hybrid mobile nodes connectivity approach for infrastructural mobile ad-hoc networks
- An open-source implementation of AODV routing protocol for Linux IPv6 ad-hoc networks

4. Conclusuions

In Deliverable D11 we have presented the plans for the verification procedure of different broadband services on the demonstration and test platforms of different BROADWAN partners. These platforms allow us to introduce the services BROADWAN will implement on different hybrid networks. The performed tests will represent the base of further evaluations.

The test sites include:

- *The demonstration site at Limoges and Paris*: This platform itself is a joint effort of CNRS and CEG. The main site is in Limoges, with broadband connection to CEG, Paris. Along with other partners (MOV, TSR, ING and TEL) we will focus on demonstrating services that generate heavy traffic. The reason of it is that most of the services (like VoD, EoD, p2p) involve video transmission. Both multicast and unicast aspects of the video transmission will be investigated.
- *The demonstration site in Spain*: It runs the demonstration for distant learning services in rural areas. The transmission is based on satellite and wireless connections. The two basis of the demonstration are the infrastructure of TCL and the IG-Class e-Learning platform from ING. An analysis is planned to be performed on video transfer over IPv6 networks.
- *The demonstration platform in Oslo*: is a 42 GHz platform created by TEL and NER. It is intended to test and verify the operation and functionality of multicast techniques using an IPv6 system.
- *The test-beds in the UK and Salzburg*: Are places where several new technologies will be tested. These technologies (like web caching, video multicast or IPv6 auto-configuration) will be essential for the future wireless networks. The main focus of research carried out by RAL, UBU and USA is on IPv6. The main goal of the test-beds is to introduce novel IPv6-based services and applications derived from the BROADWAN project.

Since BROADWAN is a collaborative project, the demonstrations and tests are prepared and will be performed as a result of exchange of experience, technology and information. There are 4 sites with about a dozen of tested services.

Finally we must also note, that these tests will also allow us to insist on the need for network interoperability, described in Deliverable D16, but also on the need for interoperability of software and application level protocols which is less evoked, but at least as relevant. The use of different platforms to implement the same services allows us to verify that solutions developed by different partners is independent from computer architecture or Operating Systems. So that these services could be offered to a larger population and also be easier to package for Service Providers.

Acronyms

ADSL	Asymmetric Digital Subscriber Line
AODV	Ad hoc On-Demand Distance Vector (AODV) Routing
BFWA	Broadband fixed wireless access
CBR	Constant Bit Rate
DNS	Domain Name Server
DVB	Digital Video Broadcast
DVD	Digital Versatile Disc
DTV	Digital television
EPG	Electronic program guide
ERP	Enterprise Resource Planning
HDTV	High Definition Television
JiST	Java in Simulation Time
IP	Internet Protocol
IPv6	Internet Protocol version 6
ISDN	Integrated Service Digital Network
LAN	Local Area Network
LMDS	Local multipoint distribution service
MAC	Medium Access Control
MAN	Metropolitan Area Network
MP3	MPEG-1 Layer III audio format
MPLS	Multi-protocol Label Switching
NAS	Network Attached Storage
NTSC	National Television System Committee
OFDMA	Orthogonal Frequency Division Multiple Access
OSI	Open Systems Interconnection
p2p	Peer-to-peer
PAL	Phase Alternating Line
PDA	Personal data assistant
PKI	Public Key Infrastructure
QoS	Quality of Service
RMP	Reliable Multicast Protocol
RRMP	Restricted Reliable Multicast Protocol
RTP	Real-time Transmission Protocol
SAP	Service Announcement Protocol
SAN	Storage Area networks
SDP	Service Discovery Protocol
SLP	Service Location Protocol
SSL	Secure sockets layer
SSM	Source-Specific Multicast
SWANS	Scalable Wireless Ad-hoc Network Simulator
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
UMTS	Universal Mobile Telephony System
VBR	Variable Bit Rate
WLAN	Wireless Local Area Network

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