# Smart PIN: Performance and Cost-oriented Context-Aware Personal Information Network


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# Declaration

I hereby certify that this material, which I now submit for assessment on the programme of study leading to the award of Doctor of Philosophy is entirely my own work and has not been taken from the work of others save and to the extent that such work has been cited and acknowledged within the text of my work.

Student ID: ______56106645______

Signed: ______________________

Date: ________________________

*To my lovely wife, to my cheerful son and to my dear parents*

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# List of publications

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**Seung-Bum Lee**, Gabriel-Miro Muntean, and Alan F. Smeaton. "User-centric Utility-based Data Replication in Heterogeneous Networks." *IEEE ICC 2008 Workshop on Digital Television and Mobile Multimedia Broadcasting*, Beijing, China, pp 290-294, 19 - 23 May 2008.

**Seung-Bum Lee**, Gabriel-Miro Muntean, and Alan F. Smeaton. "Smart PIN: Utility-based Replication and Delivery of Multimedia Content to Mobile Users in Wireless Networks." *IEEE International Symposium on Broadband Multimedia Systems and Broadcasting 2008: Mobile and Handheld Systems for Entertainment on the Go*, Las Vegas, NV, pp 1-8, 31 March - 2 April 2008.

**Seung-Bum Lee**, Gabriel-Miro Muntean, and Alan F. Smeaton. "Cost-oriented context and content data pair delivery in Smart PIN." *Proceedings of IET China-Ireland International Conference on Information and Communications Technologies (CIICT)*, Vol. 1: pp 197-204, Aug. 2007.

# Abstract

The next generation of networks will involve interconnection of heterogeneous individual networks such as WPAN, WLAN, WMAN and Cellular network, adopting the IP as common infrastructural protocol and providing virtually always-connected network. Furthermore, there are many devices which enable easy acquisition and storage of information as pictures, movies, emails, etc. Therefore, the information overload and divergent content's characteristics make it difficult for users to handle their data in manual way. Consequently, there is a need for personalised automatic services which would enable data exchange across heterogeneous network and devices. To support these personalised services, user centric approaches for data delivery across the heterogeneous network are also required.

In this context, this thesis proposes **Smart PIN - a novel performance and cost-oriented context-aware Personal Information Network**. Smart PIN's architecture is detailed including its network, service and management components. Within the service component, two novel schemes for efficient delivery of context and content data are proposed: **Multimedia Data Replication Scheme (MDRS)** and **Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD)**.

MDRS supports efficient data accessibility among distributed devices using data replication which is based on a utility function and a minimum data set. QAMMD employs a buffer underflow avoidance scheme for streaming, which achieves high multimedia quality without content adaptation to network conditions. Simulation models for MDRS and QAMMD were built which are based on various heterogeneous network scenarios. Additionally a multiple-source streaming based on QAMMS was implemented as a prototype and tested in an emulated network environment. Comparative tests show that MDRS and QAMMD perform significantly better than other approaches.

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# Chapter 1

# Introduction

## 1.1 Research Motivation

Since the 1990s, over 90% of information has been produced in digital form and this percentage is increasing [1]. As a generation of digital content is growing, the amount of information which each one of us needs to handle also increases. Personal computers have become a fundamental electronic device (e.g. people use word processors and the internet for even simple messages); many digital cameras exist (some embedded in mobile phones) which enable people to take a large number of pictures; video cameras are getting cheaper and smaller, and regular people as well as professionals are using them to make even short movie clips during their normal lives. Apart from their own user-generated data, users can also add to their own personal data via Internet-connected computers and mobile phone download services, mostly in the form of music and movies. As a result of all this, information overload [2] affects not only people working in professional scenarios but almost everybody, in our personal and in professional lives.

One of the issues which arises with having this amount of data is storage and where to store. One interesting research avenue, the MyLifeBits [3] project, tries to store a person's whole electronic data in digitised form (e.g. email, web pages, pictures, etc.) It has been estimated that the amount of data acquired is roughly 1 GB/month and 1 TB/lifetime just for one individual. Terabyte sized hard disk drives are now available in the market and can be affordable, even though the required size of storage for a person's life will very likely increase.

The problem with storing all of a person's data from their lifetime on a hard disk, is that users do not only consume their content within their own devices, but they want to share it with other people as well. Many types of files generated or residing on personal computers are already able for use on mobile devices and vice versa. For example pictures could be taken with a digital camera and they can be stored and viewed on a laptop. People may be interested to copy these pictures onto their other devices (e.g. mobile phones). They may also use social networking websites in order to share these with friends and family. In addition, some people also share files (e.g. ringtones for mobile phones) through Bluetooth connection.

Although evolving networking technologies enable these types of sharing to happen, the delivery of content is still dependent on the cost in relation to the characteristics of the network technologies used and on its different utility to their users. Conventional mobile phones have not only primary connectivity such as cellular (e.g. 3G), but also secondary network connectivity such as wireless personal area network (WPAN) (e.g. Bluetooth). Personal computers include both wired network connection and wireless local area network (WLAN) connections as default features. This heterogeneous network availability makes users virtually always-connected to the network. In addition, the latest wireless technologies for the next generation network area offer more bandwidth, support for higher mobility and lower power consumption while providing multiple connectivity technologies. However, each of these network connectivity technologies are designed for often narrow purposes and have different characteristics. For example, WPAN covers very short range, requires low power consumption, and there is no consideration of mobility. On the other hand, a Wireless Metropolitan Area Network (WMAN) supports a wide area, requires relatively high power consumption, and considers mobility issue as one of its main issues.

With the mixture of information overload and a cost-dependent heterogeneous network environment, what users want is simple and efficient usage of their content via a range of devices. For example, let us assume that there is a user (User A) who is interested in Irish dancing and music and he leaves his home with his mobile phone and digital camera to meet his friends (User B and User C) who all like Irish dancing and are interested in pop and rock music, respectively. All have stored their favourite content in their portable devices and also choose to share the data with their friends, but do not want to do it manually. They would highly benefit from automatic data exchange among uses and between devices. The infrastructure which would support such exchange is illustrated in Fig. 1.1. Possible data exchange situations are described below:

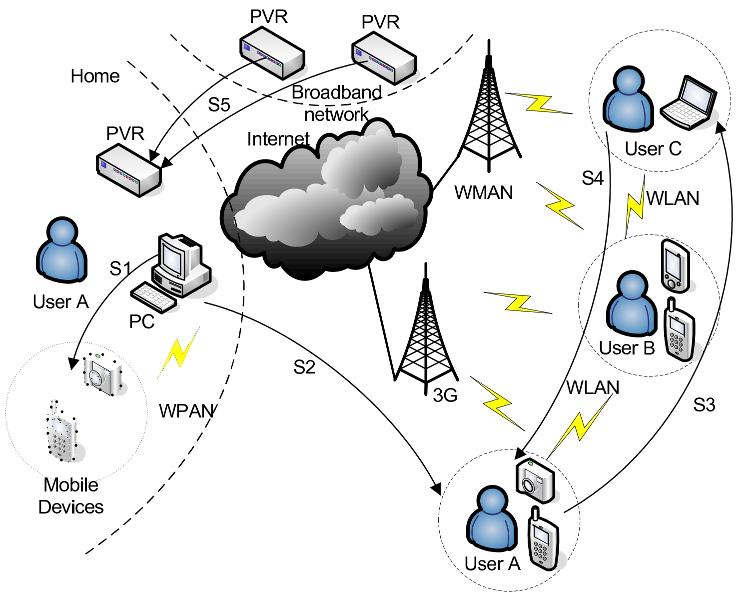



Figure 1.1: General Concept for Personal Information Network

- Situation 1 (S1): User A is at home. During breaks, he watches video clips or listens to music stored on his work PC. Based on his interest, network load, and device properties, some multimedia files are duplicated on his smart mobile phone.

- Situation 2 (S2): When User A is away from home, he takes some pictures and uploads them to his website. User A then remembers that there is some other image he wants to upload too. Some of pictures are already in his smart mobile phone.

However, there are some missing files and he can get them from his PC to mobile phone manually.

- Situation 3 (S3): User A leaves home and meets his friends, Users B and C. Based on shared interests, some data belonging to User A will be copied to the smart devices of his friends. For example, some of the Irish pop multimedia clips (e.g. Westlife) will be shared with User B and some of the Irish rock multimedia clips (e.g. U2) will be copied to User C's device. This is done according to the other users' interest as well as network load and device capabilities.

- Situation 4 (S4): Data belonging to the User B and User C will be shared with User A according to his interest, network load and device capability.

- Situation 5 (S5): When User A comes back home, he finds that he missed the U2 live transmission on TV. However, he can watch that from the start using streaming from other people's recorded copies on their Personal Video Recorders (PVRs).

The simple existing solutions such as network file systems [4] and peer-to-peer (P2P) file sharing systems [5] can only provide restricted access to desired content and users need to manually access and manipulate specific devices offering the required service. The difficulties to make service access simple are mostly due to huge amounts of data acquired and stored on distributed devices. Cloud computing storage service such as SkyDrive[1] and Google Accounts [2] [3] was also proposed in order to resolve the storing and sharing issue but there are still problems, for instance, cost variance of data delivery. Furthermore, it is not difficult to realise that services are limited to the devices accompanying users and these can provide meaningful information to users if the point of view is moved to the user. With this context, WPAN and WLAN, human-centred networks connecting devices in a spontaneous architecture within a short-rage ("personal" or "body" space) [6] are good infrastructures to

[1] Windows Live SkyDrive: http://skydrive.live.com/, last accessed 18 Nov. 2009

[2] Google Accounts: https://www.google.com/accounts/, last accessed 18 Nov. 2009

[3] Upgrading Storage : How it works, http://picasa.google.com/support/bin/answer.py?hl=en&answer=39567, last accessed 18 Nov. 2009

envisage these scenarios.

Since users cannot manage large amounts of data manually, they require an intelligent solution to solve this problem. A desirable solution is to maintain content distributed and deliver it only when it is needed. This can be considered as the current implemented system and data delivery and storage are determined based on the user's activity. However, this could lead to a set of very annoying procedures for users. Furthermore, the cost of data acquisition varies depending on its physical location. Another alternative solution is duplication of all content to mobile devices. This is not reasonable since mobile devices have limitations in terms of cost of delivery and storage. Consequently, there should be a hybrid solution between these two extreme solutions. The solution basically supports manual delivery of data and should include smart ways for automatically duplicating some content among the devices based on different users' interest levels. Since the content should be shared with different users, the dynamics caused from the hybrid data delivery solution should be taken care of during service delivery.

## 1.2 Problems and Goals

Currently available systems support automatic system configuration for heterogeneous networks, robust services adapted to changing network topologies such as Peer-to-peer (P2P) services, context-awareness with mobile devices for smart user access to information located on remote devices such as network file system and file sharing systems. Even though there are good approaches for some of the issues mentioned above, there are still remaining issues. Putting this into other words, the problems found in WPAN/WLAN-based personal information networks could be summarised as:

- Heterogeneity of available connectivity technologies which have different characteristics in the protocol layers such as the physical and MAC layers.

- Co-existence of ad-hoc and fixed networks which require different routing protocols.

- Complex and dynamic characteristics of distributed data including video, audio, picture, document, etc.
- Unstable availability of devices which move as the user moves from location to location.
- Overload of information such as personal data and public data in which the user is interested.
- Various user utilities which change across the multiple mobile devices per user.

Specifically, these issues provide various challenges in terms of performance, cost-effectiveness and quality of service supporting less user intervention. For this problem, the hybrid data delivery solution should support:

- Performance and cost-oriented storage in devices
- Efficient content delivery over cost-dependent heterogeneous networks
- User utility-orientation making use of various metrics
- Differentiated approaches based on data characteristics

## 1.3 Solution and Contributions

As a hybrid data delivery solution for the above set of problems, this thesis proposes Smart PIN, a performance and cost-oriented context-aware personal information network. Smart PIN is a novel solution that enables efficient user access to information located on adjoining and remote devices and multimedia streaming from multiple senders. Figure 1.2 presents an overview of the proposed Smart PIN. It includes a number of agents that communicate using various network technologies and provide support for information exchange. Each agent may have network connectivity with entities outside the Smart PIN space.

Smart PIN replicates data based on performance metrics, user utility and cost for data selection and delivery in order to support efficient users access. The performance metrics

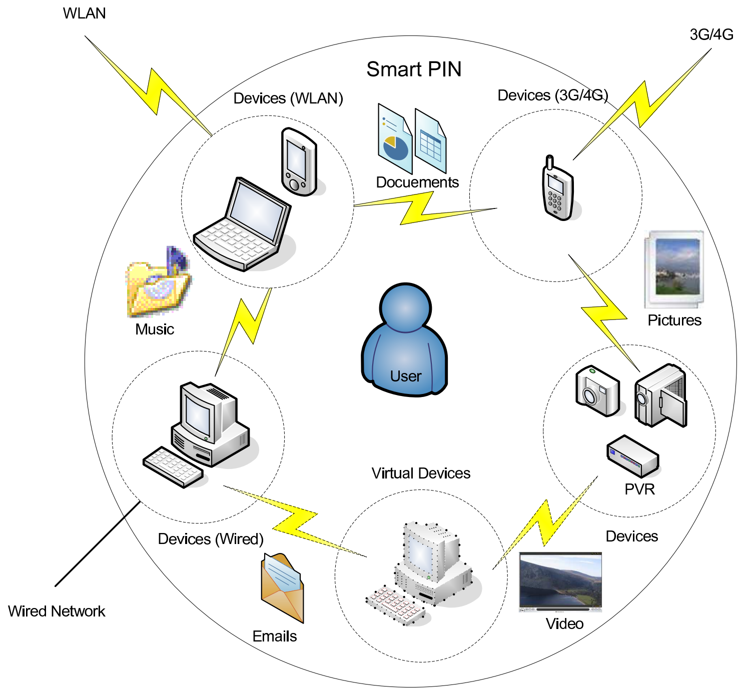



Figure 1.2: Topology of Smart PIN

include bandwidth, delay from networks, and usage of memory and other resources for devices. The cost could be network characteristics — based on things such as data rate and load conditions, data characteristic oriented parameters such as size of file, or combination of these such as throughput and delivery time involved per file, etc. Since the system is based on heterogeneous networks, the cost metrics which are used mostly will consider these aspects. User utility could address presenting how important content is for the user of a specific device. For this, metadata, information about data, should be used since it includes attributes of data during its creation and access. With this, systems can measure performance and cost metrics which could then be converted into user utility.

The replicated data includes various types such as document, pictures, video, audio, etc. In the case of video and audio, Smart PIN also supports streaming as well as downloading. Since multiple agents include the same data through data replication, multiple-senders can provide high multimedia quality, by adapting to changes in network conditions. In addition, the receiver adopts a buffer management scheme in order to achieve better quality of multimedia streaming.

As discussed above, Smart PIN originally contributes and targets the following aspects:

- A novel data replication approach for Smart PIN
  - Data selection and delivery algorithms based on user-centric utility functions resolving information overhead.
  - Minimum data set decision algorithm for data replication based on device availability.
- Multiple-source streaming with replicated data
  - A novel buffer underflow avoidance scheme supporting high multimedia quality.
- A new context-aware personal information system design
  - Three different services as a context-aware system.
  - Two novel approaches for data replication and multiple-source streaming.

## 1.4 Short Outline of the Thesis

This thesis is structured as follows. Chapter 2 discusses background technologies on the various MAC, network, transport and application layers. Chapter 3 describes related work which covers approaches, and projects related to Smart PIN and the techniques which we use. Chapter 4 introduces the proposed Smart PIN architecture. Chapter 5 discusses modelling, simulation-based tests and result analysis for Smart PIN. Chapter 6 focuses on prototyping and tests related to the multiple-source streaming part of Smart PIN. Finally, chapter 7 presents conclusions and future work.

# Chapter 2

# Background Technologies

## 2.1 Introduction

This chapter covers background technologies for Smart PIN such as networking technologies, context and metadata, data replication and transport protocols. As Smart PIN relies on networking between heterogeneous devices, the characteristics of various networking technologies are important and influence the performance of communication. Since Smart PIN is a performance and cost-oriented approach, it makes use of both context and metadata. Data replication is also discussed since Smart PIN handles distributed data among multiple devices. Understanding transport protocols is also essential, as Smart PIN performs performance-aware data delivery over network.

## 2.2 Networking Technologies

An internetwork such as the Internet is defined as a collection of individual networks which is operated as one big network [7]. Nowadays, the most popular individual networks are local area networks (LANs), for example Ethernet and Wireless LAN (WLAN), covering relatively small areas such as a home or an office. These LANs are connected through Wide Area Network technologies such as Cable network, Digital Subscriber Line (DSL), etc. Since there are many personal devices which require networking features, Personal

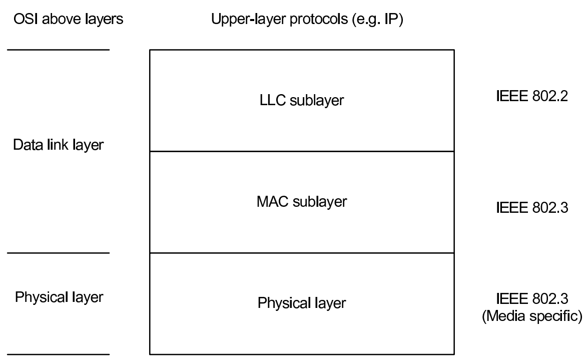



Figure 2.1: Simplified Ethernet reference model

Area Network is also emerging. In order to cover most relevant networking technologies, wired and wireless solutions are briefly discussed.

### 2.2.1 Wired Technologies

#### 2.2.1.1 Ethernet (IEEE 802.3)

Ethernet is one of the most popular local area network (LAN) technologies and it is standardised as IEEE 802.3. The draft standard was approved in 1983 and the official standard was published in 1985 as ANSI/IEEE std. 802.3-1985[1]. Four data rates are currently defined for operation over optical fibre and twisted-pair cables based on standards which are consolidated into IEEE Std 802.3-2008:

- 10 Mbps: 10Base-T Ethernet
- 100 Mbps: Fast Ethernet
- 1000 Mbps: Gigabit Ethernet
- 10 Gbps: 10 Gigabit Ethernet

In addition, 100G Gigabit Ethernet standardisation is scheduled for approval in June 2010 supporting 40 Gbps and 100 Gbps in the physical layer [8, 9].

[1] IEEE 802.3 Working Group for Ethernet, http://www.ieee802.org/3/, last accessed 18 Nov. 2009

As shown in Fig. 2.1, Ethernet includes several layers defined in the IEEE 802 standards for data link and physical layers. Logical Link Control (LLC) is the upper portion of the data link layer of the OSI Model [10]. IEEE 802.3 defines the Medium Access Control (MAC) sublayer and physical layer. The physical layer covers transmission data rate, encoding, supporting media types. Specifically, the MAC sublayer has two primary responsibilities:

- Data encapsulation, handling frame assembly for transmission and parsing for reception including error handling.
- Media access control, including initiation and failure recovery of frame transmission.

Ethernet uses duplex communication supporting the sharing of the medium using the protocol commonly known as the carrier sense multiple access collision detect (CSMA/CD). This protocol works based on distributed procedures which could be summarised as:

- Carrier sense - Each node listens to the medium in order to determine whether it can initiate transmission of frames.
- Multiple access - Any nodes may start when they detect that the network is quiet during carrier sensing.
- Collision detection - When several nodes transmit data at the same time, they can detect this data collision situation. If this happens, each node transmits again after a random length of time calculated by a back-off algorithm.

#### 2.2.1.2 Residential Broadband Technologies

Among the most known, currently available broadband technologies are Digital Subscriber Line (DSL), Cable network and Fiber-To-The-Home (FTTH) based on mature technologies [11, 12, 13]. These technologies use wire-based technologies such as cable or fibre.

DSL[2] historically started with analogue telephone lines in order to support high speed data networking. Before DSL, there was dial-up connection which uses a modem support-

[2] Broadband Forum, http://www.broadband-forum.org/, last accessed 18 Nov. 2009

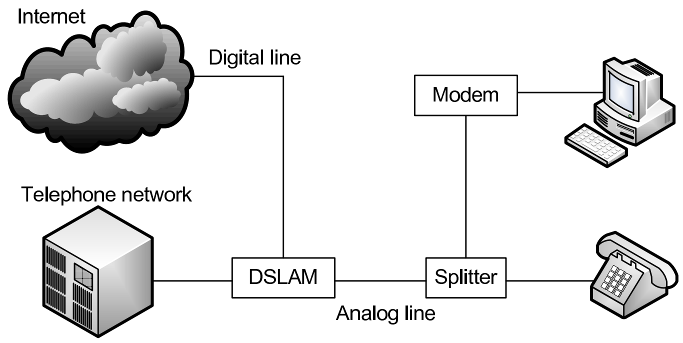



Figure 2.2: Simplified DSL network

ing several hundreds kbps. This dial-up modem shares the carrier with voice, therefore the subscriber could not use voice when the modem uses the line. Instead of that, DSL uses multiple carriers in order to support data and voice multiplexed together. In the case of Frequency Domain Multiplexing (FDM), there are three kinds of channels for high bit rate downstream data, low bit rate duplex data and basic telephone services. Fig. 2.2 shows simplified DSL network which consists of Digital Subscriber Line Access Multiplexer (DSLAM), Splitter and Modem. In a DSLAM, a high bit rate downstream and low rate duplex is merged and converted into an analogue signal. After that, DSLAM merges those digital signals with analogue voice signals and sends it to the subscriber. On the customer's side, the voice and data signals are separated through a splitter and finally the modem receives data. This multiplexing enables users to use telephone and data services simultaneously.

The standardisation activity for DSL is mostly performed by the International Telecommunication Union - Telecommunication Standardisation Sector (ITU-T) starting from Asynchronous DSL (ADSL). ADSL2 is an evolved version of ADSL in order to support higher data rates and it is currently the most popular DSL technology [11]. Table 2.1 shows that Very high bit rate DSL (VDSL) is currently the fastest commercialised technology based on DSL. In addition, there is also research on faster approaches such as Gigabit DSL (GDSL) [14] using Multiple Input and Multiple Output (MIMO) technique which is popularly adopted in various areas such as DVB-T [15]. GDSL may be possible using four

| Rec. No. | Recommendation Title | Down/Upstream Rate (Mbps) |
|---|---|---|
| G.992.1 (G.dmt) | ADSL Transceivers | 8/1 |
| G.992.2 (G.lite) | Splitterless ADSL Transceivers | 1.5/0.5 |
| G.997.3 (G.dmt.bis) | ADSL2 Transceivers | 12/1 |
| G.997.4 (G.lite.bis) | Splitterless ADSL2 Transceivers | 1.5/0.5 |
| G.997.5 (G.dmt.bisplus) | ADSL2+ Transceivers | 24/1 |
| G.993.1 (G.vdsl) | VDSL Transceivers | 52/6(26/26)[a] |
| G.993.2 (G.vdsl2) | VDSL2 Transceivers | 100/100 |

[a] Asymmetric (Symmetric)

Table 2.1: ITU-T DSL Standards

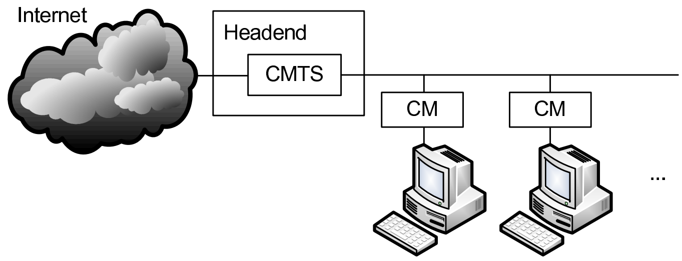



Figure 2.3: Simplified Cable network

copper twisted pairs MIMO in order to achieve 1Gbps.

Cable television (CATV) originally provided a unidirectional service delivering analogue TV channels through a coaxial cable instead of over-the-air through antenna which is used in TV broadcasting. The CATV service provider has evolved this cable network to support various services including a data service. Fig 2.3 shows a simplified architectures for cable network. The Data Over Cable Service Interface Specification (DOCSIS) [16] is the most common specification defining the interface requirements of Cable Modems (CMs).

The service provider manages a number of headends as access networks. Each CM exchanges data with a Cable Modem Termination System (CMTS) located in a headend. On the downlink, CMTS broadcasts data to all cable modems and a specific CM selects the data it needs. On the uplink, they use the time domain multiple access scheme for a CM to reserve time slots for data transmission.

FTTH[3] is a particular case of Fiber-To-The-X (FTTX). In this case, X stands for Home (or Premises). FTTX indicates which part divides the optical fibre and metallic cable in the network. Consequently, FTTH means the optical cable is used until the customer's home or premise and that optical cable is terminated with an Optical Network Terminal (ONT) as shown in Fig. 2.4. Strictly speaking, FTTH could cover all the networks with fibre, but some parts of the network still have metallic cables. One of the simple reasons for this is that many devices still need to use electrical equipment such as Ethernet cards which are still popular. In addition, VDSL is also deployed within FTTH in order to support various service configurations [17, 13].

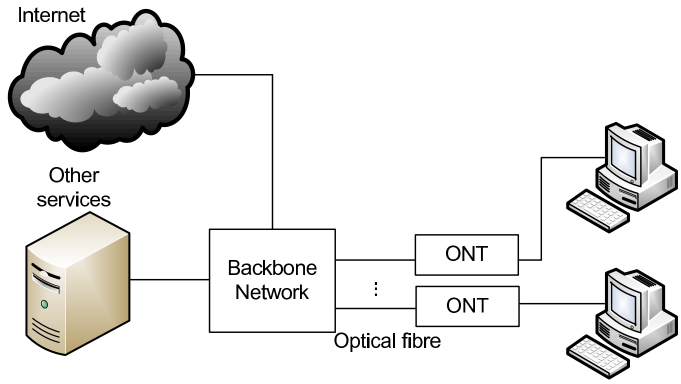



Figure 2.4: Simplified FTTH network

Optical networks used in FTTH could be categorised into Active Optical Network (AON) and Passive Optical Network (PON). AON involves switching, routing and multiplexing which is usually based on electrical equipment. On the other hand, PON just involves splitting optical signals, and supports point-to-multipoint through the fibre. Although it has limited features, enough data rate could be achieved with relatively lower costs than AON. In addition, this could be easily deployed for FTTH.

### 2.2.2 Wireless Technologies

Recently, wireless communication technologies are everywhere. Mobile phones have many interfaces such as Bluetooth, WiFi, infrared, and cellular. Laptop PC users utilise their

[3] Fiber to the Home Council, http://www.ftthcouncil.org/, last access 18 Nov. 2009

WiFi interface as well as wired for Internet connectivity. In Smart PIN, most connectivity technologies are based on wireless connections. For this reason, this section presents an introductory review of the most popular wireless standards from the IEEE including WMAN, WLAN and WPAN.

#### 2.2.2.1 WMAN - IEEE 802.16 family

Wireless Metropolitan Area Network (WMAN) standards are dealt with in the IEEE 802.16 working group[4] on Broadband Wireless Access (BWA). After their establishment in 1999, IEEE Std 802.16-2001 was approved in December 2001 and published in April 2002 [18]. Similar to other standards in the IEEE 802 working group, the IEEE 802.16 standard specifies the MAC and PHY of fixed broadband wireless access systems. Furthermore, the MAC layer is designed to support multiple physical layer specifications to give the standard flexibility. Table 2.2 shows brief descriptions of the IEEE 802.16 standard family [19].

| Name | Description |
|---|---|
| 802.16 | Fixed broadband wireless system between 10 and 66 GHz |
| 802.16a | Amendment for operation between 2 and 11 GHz |
| 802.16c | Enhancement including system profile between 10 and 66 GHz |
| 802.16.2 | Coexistence between 10 and 66 GHz |
| 802.16/Conf01<br>802.16/Conf02<br>802.16/Conf03 | Test and conformance specification |
| 802.16d | System profiles (active) |
| 802.16e | Enhancement to support mobility (active) |

Table 2.2: IEEE 802.16 standard chart

Following the understanding of the terminology of "broadband" from BWA, it is expected that WMAN systems will become a strong competitor for DSL and cable Internet connectivity [19]. The reasons are that WMAN systems are capable of delivering significantly higher data rates than competitors and support the places where customers cannot get services via DSL or cable. Furthermore, the service providers benefit from relatively low

[4]IEEE 802.16 Working Group for Broadband Wireless Access Standards, http://www.ieee802.org/16/, last accessed 18 Nov. 2009

up-front costs while consumers benefit from the convenience of a wireless connection. Currently, Irish Broadband[5] is providing WMAN services with the product name “Ripwave” in Ireland.

Although the 802.16 family is officially called WMAN, the term, WiMAX (Worldwide Interoperability for Microwave Access) created by an industry group called the WiMAX Forum[6], is used more. Additional efforts have been made in mobile WiMAX in order to achieve spectrum efficiency, less latency and wider bandwidths though 802.16m standardisation [20].

Basically, a simple WMAN topology consists of a base station (BS) and a number of subscriber stations (SS) [18]. Fixed WMAN systems have a multipoint architecture: one is point-to-multipoint (PMP) and the other is multipoint-to-multipoint (MP-MP). In PMP, routing to the appropriate BS is a function of the core network. MP-MP or mesh systems have the same functionality as PMP systems. Traffic may pass through one or more repeaters to reach a subscriber. These networks operate transparently, so users are not aware that services are delivered by radio. The range of applications includes voice, data, and entertainment services of many kinds, as other broadband services have.

This technology targets to cover much larger areas than others such as WLAN and WPAN. One of the reasons is that it uses different antenna technology. Similar to cellular technology, the sectorised antenna of the base station is capable of handling multiple independent sectors simultaneously. Furthermore, there are several groups of frequency supported which show different characteristics [19].

The first group of frequency bands for WMAN is the licensed bands between 10GHz and 66 GHz. In this band, there are two technical advantages. First, multipath is negligible. Second, the band of operation is large. This enables high data rates up to more than 120Mbps. However, thermal noise and/or interference are the main limiting factors. Rain will increase the attenuation experience. In addition, high-order modulation schemes to get high bit rate require large Signal to Noise Ration (SNR) for satisfactory operation.

[5] Irish Broadband, http://www.irishbroadband.ie/, last accessed 18 Nov. 2009
[6] WiMax Forum, http://www.wimaxforum.org/node, last accessed 18 Nov. 2009

The second group of frequency bands for WMAN is the licensed bands between 2GHz and 11 GHz. Line of sight is not necessary. Multipath can be significant and appropriate measures must be taken. To take care of this issue, advanced power management, automatic repeated request (ARQ), and space time coding (STC) is adopted. The third group of frequency bands is the unlicensed bands between 2GHz and 11GHz. This band shows the same physical characteristics as licensed bands. Since these bands are unlicensed, users who could cause interference may exist. Second, regulation limits the output power. For these issues, Dynamic Frequency Selection (DFS) and power management is required.

In the view of protocol layers, MAC includes a convergence sublayer, common part sublayer, and security sublayer [18]. The convergence sublayer is for better handling of the higher-layer protocol placed above the MAC. The central part of MAC is the common part sublayer. It handles channel access, connection establishment and maintenance, and QoS. The security sublayer provides authentication, secure key exchange, and encryption. Since 802.16 uses a framed PHY, the MAC aligns its scheduling intervals with the underlying PHY framing.

The upstream physical layer is based on the use of a combination of time division multiple access (TDMA) and demand assigned multiple access (DAMA). The upstream channel is divided into a number of ”time slots”. The downstream channel can be either based upon continuous time division multiplexing (TDM) or burst mode of operation. Continuous TDM multiplexes the information for each SS onto the same stream of data. All SSs located within the same sector receive that data. In the case of burst mode of operation, the way is similar to the way of TDMA upstream bursts. This allows bursts to be transmitted to specific SSs.

Table 2.3 gives a summary of the different WMAN systems [19]. For the duplexing of upstream and downstream, Frequency Domain Duplexing (FDD) and Time Domain Duplexig (TDD). FDD employs a different frequency for each stream, whereas TDD assigns different time slots for each stream. In addition, AAS in Table 2.3 stands for Advanced Antenna System which includes techniques such as Multiple Input and Multiple Output

(MIMO) and Beamforming.

| Standard | Band | PHY | MAC | Duplexing |
|---|---|---|---|---|
| WMAN-SC | 10-66 GHz | SC | Basic | TDD,FDD |
| WMAN-SC2 | 2-11 GHz | SC2 | Basic+ARQ+STC+ AAS | TDD,FDD |
| WMAN-OFDM | 2-11 GHz licensed | OFDM | Basic+ARQ+STC +DFS+AAS | TDD,FDD |
| WMAN-OFDM | 2-11 GHz unlicensed | OFDM | Basic+ARQ+STC +DFS+mesh+AAS | TDD |
| WMAN-OFDMA | 2-11 GHz licensed | OFDMA | Basic+ARQ+STC +DFS+AAS | TDD,FDD |
| WMAN-OFDMA | 2-11 GHz unlicensed | OFDMA | Basic+ARQ+STC +DFS+mesh+AAS | TDD |

Table 2.3: IEEE 802.16 short summary

#### 2.2.2.2 WLAN - IEEE 802.11 family

Like other IEEE 802.11 family[7], Wireless Local Area Network (WLAN) standards supports higher bandwidth and has better usability in comparison with other wireless technologies. IEEE 802.11 standards define media access control (MAC) in layer 2 and the physical layer. Table 2.4 and table 2.5 show brief characteristics of the IEEE 802.11 standard family [19].

| Standard | Year | Frequency | BW for PHY[a] | BW for MAC SAP[b] |
|---|---|---|---|---|
| 802.11 | 1997 | 2.4 GHz | 2 Mbps | - |
| 802.11b | Jul. 1999 | 2.4 GHz | 11 Mbps | 5 Mbps |
| 802.11a | Jul. 1999 | 5 GHz | 54 Mbps | 25 Mbps |
| 802.11g | 2002/2003 | 2.4 GHz | 54 Mbps | 25 Mbps |
| 802.11n | Nov. 2009 | 2.4/5 GHz | 600 Mbps | 144 Mbps |

Table 2.4: IEEE 802.11 standards short summary

[a] BW: Bandwidth; PHY: Physical layer;
[b] SAP: Service Access Point

In IEEE 802.11 MAC, three configurations for network topology are supported [21]. The first is Independent Basic Service Set (IBSS) which enables devices to form a network

[7] IEEE 802.11 Working Group Setting the Standards for Wireless LANs, http://www.ieee802.org/11/, last accessed 18 Nov. 2009

| Name | Description |
|---|---|
| 802.11 | The original WLAN Standard. Supports 1Mbps to 2Mbps |
| 802.11a | High speed WLAN standard for 5GHz band. Supports 54Mbps |
| 802.11b | WLAN standard for 2.4 GHz band. Supports 11 Mbps |
| 802.11d | International roaming - automatically configures devices to meet local RF regulations |
| 802.11e | Addresses quality of service requirements for all IEEE WLAN radio interfaces |
| 802.11f | Defines inter-access point communications to facilitate multiple vendor-distributed WLAN networks. |
| 802.11g | Establishes an additional modulation technique for 2.4Ghz band. Supports speeds up to 54 Mbps |
| 802.11h | Defines the spectrum management of the 5 GHz band. |
| 802.11i | Addresses the current security weaknesses for both authentication and encryption protocols. The standard encompasses 802.1x, TKIP, and AES protocols |
| 802.11n | Provides higher throughput improvements. Intended to provide speeds up to 500 Mbps. |

Table 2.5: IEEE 802.11 standard chart

without an access point (AP), communicating among themselves in ad-hoc manner. The second is Basic Service Set (BSS). In this configuration, there is an AP which is connected to the wired network. All other devices connect to the network infrastructure through this AP. The Extended Service Set (ESS) indicates the topology which has two or more BSSs. Each BSS establishes an independent network and those form a single subnet separately. Furthermore, there are defined two modes of operation: ad hoc mode adopted by IBSS and infrastructure mode utilising AP based on BBS and ESS topology.

IEEE 802.11 physical layer can use Diffused Infrared and Frequency Spread Spectrum [21]. In Frequency Spread Spectrum, there are two types of modulation schemes such as Frequency Hopping Spread Spectrum (FHSS) and Direct Sequence Spread Spectrum (DSSS). With FHSS, radio design can be less complex, but DSSS is less susceptible to radio noise and creates little interference. IEEE 802.11b supports data rates as indicated in table 2.6 [22]. To support those various physical characteristics and separate MAC dependent part, the sub-layers in the physical layer include Physical Layer Convergence Protocol (PLCP) and Physical Medium Dependent (PMD) sub-layers.

| Data Rate | Code Length | Modulation | Symbol Rate | Bits/Symbol |
|---|---|---|---|---|
| 1Mbps | 11 (Barker Seq.) | BPSK | 1MSps | 1 |
| 2Mbps | 11 (Barker Seq.) | QPSK | 1MSps | 2 |
| 5.5Mbps | 8 (CCK) | QPSK | 1.375MSps | 4 |
| 11Mbps | 8 (CCK) | QPSK | 1.375MSps | 8 |

Table 2.6: IEEE 802.11b data rate specification

| PLCP preamble | | PLCP header | | | | Payload |
|---|---|---|---|---|---|---|
| Synchronization (128bits) | SFD (16bits) | Signal (8bits) | Service (8bits) | Length (16bits) | HEC (16bits) | Payload (Variable) |

Table 2.7: IEEE 802.11b DSSS PHY frame format

Basically, IEEE 802.11 MAC is similar to Ethernet, defined for wired networks. However, IEEE 802.11 MAC supports following features which are different [21]:

- Authentication: Wired Equivalent Privacy (WEP)
- De-authentication: Process of denying client credentials
- Association: Establishment of wireless link between Client and AP
- Disassociation: Cancelling wireless link
- Re-association: Addition to association when a wireless client moves from one BSS to another
- Privacy: WEP option encrypts data using a 40-bit encrypting algorithm known as RC4
- Data Transfer: Collision Sense Multiple Access with Collision Avoidance (CSMA/CA)
- Distribution: The distribution system (DS) during frame transmission between APs.
- Integration with portal: Logical integration between existing wired LANs and 802.11 LANs which is similar to a bridging function in a wired network.
- Power management: Active Mode (Wireless Client is powered to transmit and receive) and Power Save Mode (Client is not able to transmit or receive, consuming less power)

Ethernet uses a medium access mechanism, Collision Sense Multiple Access with Collision Detection (CSMA/CD) which is very effective in a wired environment. However, users experience significant performance degradation if there are 30-40% of collisions [23]. To prevent the potential conflicts in a shared medium, the following schemes are used for wireless environments: the first is negotiation of the data exchange before the collision happens, the second is forcing non-active users to defer their translation for a period of waiting time.

There are two reasons for CDMA/CD not to be implemented for WLANs [22]. The first is the hidden node problem. WLAN equipped nodes hear each other. However there exist hidden nodes which hear the AP, but may not hear some other nodes associated with AP. The second problem is that WLAN uses half duplex. In other words, a node cannot both transmit and receive at the same time. Instead of CDMA/CD, IEEE 802.11 adopts Collision Sense Multiple Access with Collision Avoidance (CSMA/CA) supporting the following features:

- Point Coordination Function (PCF): For time-critical services, AP polls clients who can send data and assign a time slot for a node to use. This is an optional feature.
- Distributed Coordination Function (DCF): A sender senses the medium first if it has data to send. If the medium is busy, then it waits and defers the transmission. If the medium is free for a period of time defined as Distributed Inter Frame Space (DIFS), the sender sends the data. After receiving data, the receiver sends ACK. If the sender does not receive an ACK, it retransmits the last fragment again.

#### 2.2.2.3 WPAN - IEEE 802.15 family

There have been various proposals with different technologies for Wireless Personal Area Network (WPAN) in order to support communication among devices close to a person. When IEEE started standardising WPAN technologies, the IEEE 802.15 working group[8] proposed the following descriptions as shown in Table 2.8. It is interesting to note that the

[8] IEEE 802.15 Working Group for Wireless Personal Area Network, http://www.ieee802.org/15/, last accessed 18 Nov. 2009

Bluetooth specification came first and the IEEE 802.15.1 standard was released later. As shown in table 2.8, IEEE 802.15.2 deals with interference of 802.11 WLAN and 802.15.1, Bluetooth because they share the 2.4 GHz band [6]. The ZigBee[9] specification is a suite of high level communication protocols based on IEEE 802.15.4. Since IEEE 802.15.4 and ZigBee are targeting a slow data rate for sensor networks, this thesis does not discuss those in detail.

| Standard | Description | Popular name |
|---|---|---|
| 802.15.1 | Identical to Bluetooth | Bluetooth |
| 802.15.2 | Dealing with interference between WLANs and PANs | |
| 802.15.3 | Higher data rates ad hoc networks | UWB |
| 802.15.4 | Lower data rate and lower cost versions for sensor networks | Zigbee |

Table 2.8: IEEE 802.15 standards short summary

Currently, a matured technology for WPAN is Bluetooth[10]. IEEE 802.15.1 is a standard for Bluetooth which is actually a specification. Bluetooth utilises Frequency Hopping Spread Spectrum (FHSS) providing up to 1 Mbps in the unlicensed 2.4-GHz band and it covers a short-range from 10 cm to 10m. Bluetooth establishes a piconet which is a set of devices and consists of one master and several slaves using a star network topology. These piconets may be combined to form a scatternet through a device acting as a master device in a piconet and a slave device in another. The master device is in charge of synchronisation, which is important for FHSS.

The target application of Bluetooth 1.0 was for a universal low-cost, wireless replacement of Universal Serial Bus (USB) and serial cables. In addition, Bluetooth devices communicate via encrypted links to provide covert communication. With Bluetooth 1.1, they tried to solve the initial interoperability problem among Bluetooth 1.0 devices. The latest version, Bluetooth 2.0 will be able to transfer up to 20 Mbps in ranges of up to 50m.

In the IEEE 802.15 working group for WPAN, IEEE 802.15.3 is for higher data rates than other standards. The IEEE 802.15.3 standard supports data rates from 11 to 55 Mbps.

[9] ZigBee Alliance, http://www.zigbee.org/, last accessed 18 Nov. 2009
[10] Bluetooth SIG, http://www.bluetooth.com/bluetooth/, last accessed 18 Nov. 2009

Furthermore, there were efforts in order to get low power, high data rates and short range with new technologies such as Ultrawide Band (UWB), which promises to be a technology for future WPAN.

The literature from Wylie-Green et al. [24] provides some details on spectrum regulation, and physical/MAC layer details based on a multi-band OFDM UWB (MB-OFDM UWB) solution. Furthermore, power saving characteristics of the MAC layer are discussed. This includes decentralised network formation, less activity in idle mode and hibernation functionality with more than two devices and power saving in between the fragmentary data transfer even if it is supporting high throughput. However, there are very few descriptions of QoS support and physical layer rate control schemes.

Currently, IEEE 802.15.3a, the WPAN addendum for a new physical layer using UWB, is stalled because of selection of multiple access technologies between Direct Spreading UWB (DS-UWB) from the UWB Forum[11] and MB-OFDM UWB from WiMedia[12]. However, WiMedia gets approval for standardisation through ECMA [25] supporting data rates from 53.3 to 480 Mbps. Moreover, there are more activities such as integration with the current Bluetooth specification and so on. The application of UWB will be not only the replacement of wired connections such as Wireless USB[13] but also IP connection between mobile devices through the WiMedia Network (WiNet).

There is another alternative physical layer based on the 60GHz band, so called mmWave WPAN [26]. This technology is currently involved with standardisation as 802.15.3c. Coexisting all other microwave systems in the 802.15 WPAN family, mmWave WPAN uses the unlicensed 57-64GHz band. Very high data rates over 2 Gbit/s are targeting services such as high speed Internet services, HDTV content delivery, etc. Wireless HD [14] is the leading industrial consortium for HD TV applications using this technology.

[11] UWB Forum, http://www.uwbforum.org/, last accessed 18 Nov. 2009
[12] WiMedia Alliance, http://www.wimedia.org/en/index.asp, last accessed 18 Nov. 2009
[13] USB.org - Certified Wireless USB, http://www.usb.org/developers/wusb/, last accessed 18 Nov. 2009
[14] Wireless HD, http://www.wirelesshd.org/, last accessed 18 Nov. 2009

### 2.2.3 IPTV: a Service Example in Networking

Internal Protocol TV (IPTV) is one of the emerging services for broadband service providers and it requires large bandwidth. Worldwide, a number of operators and companies are involved in IPTV standardisation through the Alliance for Telecommunications Industry Solutions (ATIS), International Telecommunication Union Telecommunication Standardization Sector (ITU-T), Digital Video Broadcasting (DVB), etc. [27]. These propose standards which focus not only on the usual service-related aspects such as portability, scalability, interoperability, performance and accounting, but also on content protection and architectural elements. There are various architectural models of an IPTV distribution network, however, the architecture presented in Fig. 2.5 is commonly accepted as a typical one [28, 29].

Fig. 2.5 illustrates how the IPTV backbone network includes the super head end (SHE) and a number of video hub offices (VHOs). Both of these support encoding and packetising of broadcast TV channels and transmit through an IP network, though SHE usually covers global video content and VHO handles local video content. Both of these include VoD servers which store movies, cache broadcasted shows, etc. A VHO feeds its content to a number of video service offices (VSOs) located in the backbone network of the service provider. Finally, VSOs, which are attached to access networks such as DSL, cable, fibre, wireless, etc, distribute content to users' equipment which could be STBs, modems and/or home-gateways (HGs).

IPTV standards define many aspects in relation to the IPTV distribution, but there is no intention to restrict the market development and the innovation and creativity of researchers and developers. These standards define multimedia services for live TV content distribution and on-demand services over IP-based networks with a quality level equivalent to conventional TV. However, there are rare specific descriptions or restrictions on novel interactive and proactive services.

The Open IPTV Forum [15] defines architecture and user-network interfaces (UNIs) for Open Internet and Manager networks such as broadband based IPTV service providers [30].

[15] Open IPTV Forum, http://www.openiptvforum.org/, last accessed 18 Nov. 2009

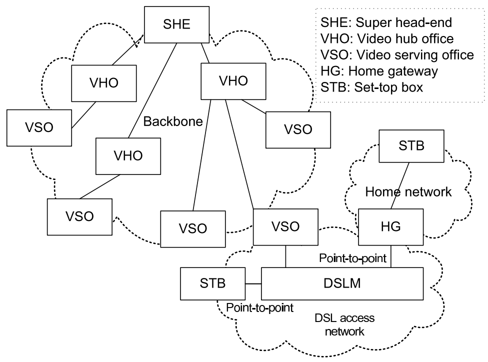



Figure 2.5: A typical IPTV network architecture with DSL access network

Since they also try to invite 3rd party content providers as a part of service providers, the service should be based on a server-client architecture and there is no consideration of peer-to-peer based content sharing among the users.

### 2.2.4 Summary

This section presented the important characteristics and some details of physical and MAC layers for major IEEE standards for wired (e.g. Ethernet and some broadband technologies) and wireless (e.g. WMAN, WLAN and WPAN) networking technologies. Following these differences, Smart PIN needs to consider these characteristics for adapting and choosing efficient usage of connections.

## 2.3 Context and Metadata for Personal Information

Usually, understanding of the history and background on the previous events helps people work better. This is also true for the computing systems with context and metadata providing important information to users. As computers usually just follow the instructions that a user provides, it is also very useful if there is more information on data which includes not only the related environmental information of the data but also information on the data itself. Usually, **context** is the information from the environment which is available to a computing system, so called context-aware system. The most popular context used in applications is user location [31, 32]. There could be other information such as time and occasion [33], objects such as places, people as well as things [34]. **Metadata** is defined as data about data. In other words, metadata includes information about the content which may be a picture, a video or an audio sequence. In addition, metadata could include file attributes such as the owner, author, creation date, etc. Consequently, context is a good source of metadata. For example, location and time when content is generated could be useful for later usage as metadata. In this section, a short discussion on context-aware systems and automatic metadata annotation will be presented for understanding of the relationship between context and metadata.

One good explanation of what context-aware systems are comes from the work of Baldauf and Dustdar [35]: "Context-aware (or sentient) systems are able to adapt their operations to the current context without explicit user intervention and thus aim at increasing usability and effectiveness by taking environmental context into account." The same authors also introduce the major issues in relation to context-aware systems such as architecture, data sensing, developing a usable context model, context processing, resource discovery, the management of historical context data, security and privacy. Most context-aware systems define at least three architectural layers: sensor, middleware, and application. In the sensor layer, there are physical sensors, virtual sensors and logical sensors. **Physical sensors** are those that actually sense the environment and acquire data. **Virtual sensors** are adopted when the context comes from virtual sources such as applications. **Logical sensors** use physical and virtual sensors and combine additional information from databases or various other sources in order to solve more complex tasks [36].

In middleware, some context-aware systems employ more layers for aggregating, processing and interpreting sensor contexts for applications. Applications utilise middleware interfaces for specific purposes. Recent research on context-aware systems is concerned with the framework in the view of system architecture [37, 38, 39], and metadata and ontologies in the view of a context model [40, 41, 32], for the general purpose context-aware system. There are also several approaches focusing on a specific topic such as user, network and activity to achieve their different system goals [42, 43, 44, 45].

As applications of context-aware systems, smart environments build context-aware systems on a specific physical place which requires users to move within its neighbourhood such as the home, at work, in the car, etc. [46]. Examples of smart environments include EasyLiving[16], House_n[17], and Interactive Workspace[18]. Most of them not only enhance computing systems, but also introduce special hardware for providing context-awareness to the existing physical items. However, it is not difficult to realise that services are limited to the user device and only devices near to the user can provide meaningful information for the user if the point of view is moved towards that user.

Other application of context-aware systems use automatic metadata annotation, a required approach for future multimedia applications, because content-based retrieval [47, 48] and semantic retrieval of multimedia [49] have been popular fields for multimedia researchers. Since the collection of metadata has been criticised because of user's difficulty of manual annotation, Davis [50] introduces an automatic multimedia metadata annotation approach which uses information from a context-aware system as a part creating metadata for content. Moreover, they categorised context into spatial, temporal and social information. In their second prototype, MMM2 [32], they showed how Bluetooth could be related to social context and other features such as automatic background picture uploading and sharing with web-based applications.

There are several standards for presenting metadata such as MPEG-21[19] and Resource Description Framework (RDF)[20]. The MPEG-21 Multimedia Framework enables the use of digital items consisting of resource, metadata and structure across a wide range of networks and devices [51]. The scope of MPEG-21 is focused on processing of metadata as glue for existing technologies. Furthermore, there are additional activities which are defined as

[16]Microsoft Easy Living, http://research.microsoft.com/easyliving/, last accessed 18 Jan. 2007

[17]MIT House_n, http://architecture.mit.edu/house_n/, last accessed 18 Nov. 2009

[18]Stanford Interactive Workspace, http://hci.stanford.edu/research/istuff.html, , last accessed 18 Nov. 2009

[19]MPEG-21: http://www.chiariglione.org/mpeg/standards/mpeg-21/mpeg-21.htm, last accessed 18 Nov. 2009

[20]Resource Description Framework (RDF)/ W3C Semantic Web Activity: http://www.w3.org/RDF/, last accessed 18 Nov. 2009

separate parts of MPEG-21 such as scalable video coding (SVC. Part 13), conformance testing (Part 14), event reporting (Part 15), binary format (Part 16), fragment identification for MPEG Media Types (Part 17), and Digital Item Streaming (DIS. Part 18).

RDF is a markup language which can present and enable to access all web resource's metadata such as Uniform Resource Identifier (URI), title, author and so on [52]. XML[21] can be considered as an alternative but RDF is more flexible and extensible in the view of formality. Since the types of most properties are text, the required performance for processing is expected to be very low. Even if the data which is needed to be gathered is huge, bandwidth for that should be quite small.

In conclusion, metadata could be good containers for the context information about the computing environment of the user. Furthermore, metadata could be good tools for the search function for information since it enables data localisation, data assessment, data selection, and data retrieval [53]. In addition, the application of metadata will not be restricted to web search since other applications may benefit from the features which can be gained from metadata. In this context, Smart PIN is also required to adopt context and metadata for autonomous handling of contents.

## 2.4 Data Replication in Distributed Data System

### 2.4.1 Information Retrieval Systems for Peer to peer network

Peer-to-peer (P2P) networking [54] is overlay networking which consists of peers rather than using classic client-service based networking and supports various features such as peer search, routing, content search and content delivery, etc. These peers have similar role in order to cooperate with each other rather than act as a server or a client.

Because of their scalability, fault-tolerance and self-organising features, peer-to-peer (P2P) systems are getting more popular for establishing large information systems. As the data processed in the system grows, the system requires more advanced approaches such as

[21] Extensible Markup Language (XML): http://www.w3.org/XML, last accessed 18 Nov. 2009

data indexing, ranking of information, etc. In this context, we will discuss the Information Retrieval (IR) approaches adopted in peer-to-peer networks.

The work of Tang et al. [55] focuses on building an Information Retrieval (IR) System on a P2P network. The basic ideas behind this P2P IR is combining IR data indexes and a P2P network over Cartesian space based on a Vector Space Model (VSM) and Latent Semantic Indexing (LSI) in the view of IR systems and Context-Addressable Networks (CANs) in the view of the P2P network. The biggest issue for collision of these concepts is mismatching of the dimensionality of LSI and the CAN Cartesian space. To solve this issue, they propose Rolling-Index. They also propose schemes for balancing distributed index and search heuristics. They simulated a P2P configuration for testing using the TREC[22] corpus for documents. For the indexing, they adopted Cornell's SMART system [56]. The metrics they used for the test are visited nodes during a search and accuracy of search results they define in the paper. With the proposed algorithms, they show the system is insensitive to major parameters such as number of nodes, rotated semantic spaces, size of the sample set, etc.

Because of the complexity of the system, the approach of Tang et al. is not popular. Instead, Breadth-First Search (BFS) is a more popular algorithm which was adopted by Gnutella P2P file sharing [57]. This simple algorithm makes a node receive the query and forward that to other nodes except the sender. The problem of this algorithm is network overhead. To reduce this, there are several proposals such as Random BFS (RBFS) [58], and Random Walker Searches [59], but their accuracy decreases.

With the aim of reducing communication costs, Intelligent Search Mechanism (ISM) [58] also tries to do this with two components: profile mechanism and relevance rank. A node in ISM stores the recent query and query hits to decide the next query. Furthermore, a node measures the relevance rank of neighbours to determine which node will get the query. A problem with this system is that a newly joined node could not get the query at all. For this, they also proposed a small random subset of peers and put it into relevant peers for

[22]Text REtrieval Conference (TREC), http://trec.nist.gov/, last accessed 18 Nov. 2009

each query.

In summary, the main challenge for P2P IR is "building overlay topologies in which close-by nodes have semantically related documents and interests" [57]. With P2P IR, they make use of the benefit of multiple data facsimiles such as enhancement of performance with a smaller number of visited nodes [55] and extended data availability [60]. This is still valid in a mobile environment if a P2P system is used on top of that. Moreover, the unstable device availability requires dynamic adaptation with data replication. In the next section, data replication and caching will be compared. After that, previous data replication systems will be outlined.

### 2.4.2 Data Replication vs. Data Caching

Sometimes, data replication and data caching are considered the same [61]. However, they show different characteristics depending on system architecture. Fig. 2.6 depicts that the data caching scheme assumes that there is a server which has all data. Usually, this could be considered as a separate server outside of the ad-hoc system. This server provides services such as web, email or file system. Other nodes are considered as clients accessing the server to query whether data is located in the server and retrieve the required data. For this scenario, the data cache scheme enables nodes on the routing of query and data transport to cache information on the data (this could be only location data or even actual data) and to use it later.

Yin and Cao [62] proposed a cooperative caching scheme for ad hoc networks. They assumed that the system has a server providing all data, and replication might show lower performance because of initial distribution overhead and intermittent connectivity. Cooperative caching which is originally for wired technologies allows the sharing and coordination of cached data among multiple nodes. In this work, they proposed CacheData (caching the data in nodes between source and destination during transfer), CachePath (caching the nearest location of data during data transfer), and HybridCache (heuristic switching algorithm of CacheData and CachePath). They compared their proposed schemes with simple ap-

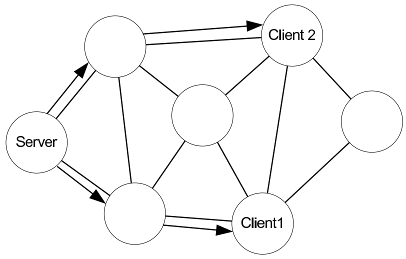



Figure 2.6: Data caching scheme

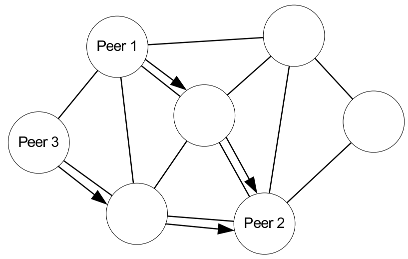



Figure 2.7: Data replication scheme

proaches and show that the HybridCache has better performance in query time, cache hit ratio in most of overall conditions.

In contradiction, data replication scheme assumes devices as peers which have a similar capability. These peers hold the data which could be generated by the user. This system also utilises the query and retrieve the scenario for data transfer. During this process, peers can store information, usually actual data. This is called the data replication scheme as presented in Fig. 2.7.

### 2.4.3 Introduction to Data Replication Systems

#### 2.4.3.1 Categorisation based on system architecture

As discussed earlier, data replication is an essential feature of the application of mobile P2P systems, and one of the most promising approaches for the computing environment. Even though there are several different approaches for data replication, in this section we will discuss three categories of data replication systems in terms of their system architecture.

**Server-based approaches** such as OmniStore [63, 64] and OceanStore [65] include a server which contains replicated data and clients wanting to access it. In some systems such as OmniStore, clients are also involved in data replication. However, some systems such as OceanStore do not let clients replicate data.

Sharing systems for mobile users such as Bayou [66] support a flexible server-client scheme which is closer to peer-to-peer systems. Bayou adopts a **distributed architecture** and supports active read and write operations. It also supports a read-any and write-any

(RAWA) replication scheme. However, these requirements cause consistency and conflict issues for every procedure which involves read and write requests. For this, they utilise various techniques such as the anti-entropy protocol, session based approaches which could be considered as overhead.

There is also a **hybrid model** such as Roam [67]. Roam elects master devices which create connection points between wide area replication domains (WARDs) which is the group of mobile devices and manages the total data set in WARD. Even though there are master devices, they just relay the requests instead of aggregating other devices and the system changes the master continuously.

#### 2.4.3.2 Categorisation through data accessing interface

There could be different approaches in terms of the data accessing interface i.e. the file system: a traditional hierarchical file system interface or a semantic file system interface.

The hierarchical file system interfaces assume that the system manages the volume, directory and files for the data. With this approach, the user needs to provide a path for the location of data to access it. Distributed data replication systems such as Roam [68] adopts this interface type. The semantic file system [69] interface provides query and response for the accessing system. Files in a semantic file system are interpreted to produce metadata which is a set of descriptive attributes. That metadata enables retrieval of the files through the queries later. OmniStore [63, 64] is a system which adopts this interface type. Since semantic file systems also assume to support a hierarchical file system through virtual directories, there are many systems which are difficult to uniquely categorise such as Wayfinder [70]. In case of EsemBlue [71], they mention the virtual directory of the semantic file system, but there is no description of queries for data accessing for a user.

### 2.4.4 Summary

In this section, a short discussion on the necessity of data replication in P2P applications such as IR system was presented. In the literature, the importance of replication for large scale information systems is emphasised, especially for mobile device based systems. We presented a comparison of data replication and caching, and a classifications of data replication systems based on system architecture and accessing interface. Other detailed issues for data replication systems will be presented in the related work section.

## 2.5 Transport Protocols

### 2.5.1 Introduction

Smart PIN relies on Internet Protocol (IP) for network layer support, and therefore various transport protocols based on IP will be used. The traditional IP transport protocol, User Datagram Protocol (UDP) [72] and Transmission Control Protocol (TCP) [73] will be introduced and their limitation will be presented. Real-time Transport Protocol (RTP) [74] is actually an application level protocol, but it is also an important protocol for multimedia delivery over IP network. After that, a short discussion on the concept of congestion control will follow. Based on this knowledge, new generation transport protocols such as Streaming Control Transport Protocol (SCTP) [75, 76] and Datagram Congestion Control Protocol (DCCP) [77] are introduced.

### 2.5.2 UDP and TCP

UDP [72] is non-reliable connection-less transport protocol which works over IP. In order to achieve this, UDP adopts a very simple header including port of source and destination, checksum and no control field. Because of this simple header, there is no support for reliability. Consequently, there might be packet ordering problems and loss of packets during delivery. However, this simplicity also provides freedom of control in the view of application layer. For example, UDP is used for many application level protocols such as Domain Name Service (DNS), Dynamic Host Configuration Protocol (DHCP) and multimedia streaming, supporting various algorithms.

TCP [73] is reliable connection-oriented transport protocol which works over IP. TCP is targeting for accurate delivery and it occupies large parts of Internet traffic [78]. In order to achieve its goal, it has procedures for connection setup and disconnection using a handshake mechanism. Between these two procedures, actual data delivery is performed. Since IP is assuming a packet-switching protocol, there could be unpredictable loss of data, arrival of data out of sequence, network congestion, etc., and TCP should cope with these.

During the data delivery, loss of data in the network can occur. There are two possible reasons: the first one is an error in the packet and the other is packet loss. For the internal packet error, a checksum is used and the packet is discarded if there is error. TCP can detect packet loss with sequence number and use acknowledgements for retransmission. In addition, the sequence number is used for reordering of packets. Due to these features, TCP has too much overhead for multimedia streaming.

There is usual confusion between the two terms “flow control” and “congestion control”. Flow control is actually related to an end-to-end relationship but congestion control concerns the network in the middle. Flow control could be performed controlling the sending rate based on the receiver’s capability. In the case of TCP, sliding windows are used for this issue. Congestion control tries to control the amount of data in the network in order to avoid congestion collapse [79, 80]. There are several mechanisms for this and those are majorly to control the rate of traffic from the sender such as Additive Increase/Multiplicative Decrease (AIMD) scheme.

### 2.5.3 Protocols related to Real-time Data Delivery

RTP [74] is proposed for delivering multimedia data through the IP network. However, it does not actually support timely delivery of multimedia data. Although RTP looks like a transport protocol, it actually uses other transport layer protocols such as UDP. Therefore, it is considered as an application layer protocol. In [74], Real-time Transport Control Protocol (RTCP) is also defined in order to control data delivery over RTP. RTP and RTCP use different port numbers which are even and odd respectively. RTP delivers multimedia data itself and RTCP delivers control packets which include information on throughput, loss, jitter, etc. This information is not used by RTP, but it is usable by the application directly. In this context, RTP can not guarantee the Quality of Service (QoS) at all.

Real Time Streaming Protocol (RTSP) [81] is a protocol which enable a client to have the features which makes multimedia streaming server similar to a VCR. The features could be play, stop, pause, etc. It is usually used in conjunction with RTP for delivering multimedia data.

### 2.5.4 Congestion Control Protocols

As discussed in the TCP section, congestion control is one of the approaches for avoiding congestive collapse in the network [79, 80]. Since TCP is commonly used in the Internet, other connection-less protocols also need to compete fairly with TCP. Due to this TCP-friendliness issue, there are several proposals for TCP friendly congestion protocols such as Rate Adaptation Protocol (RAP) [82], TCP Friendly Rate Control (TFRC) [83], etc.

Rate Adaptation Protocol (RAP) [82] is based on a similar congestion avoiding scheme used in the TCP protocol, the so-called Additive Increase/Multiplicative Decrease (AIMD) scheme. Because of its characteristics, the rate adaptation of RAP may show best performance of similar rate control approaches in terms of TCP-Friendliness. However, the rate change could be too extreme for multimedia delivery.

TCP Friendly Rate Control (TFRC) [83] is considered as an equation based congestion control protocol. It uses the simple model of TCP throughput which is represented with loss and round trip time. A receiver estimates this throughput and reports it to the sender. Based on this, the sender decides the transmission rate. Since TFRC provides throughput estimation of connection between server and client, there are many applications [84, 85, 86, 87] using this rate estimation including network abstraction.

### 2.5.5 SCTP and DCCP

There are several limitations in TCP, as discussed in [75]. Most important one of those implies that TCP and UDP are two opposite ends of the scale. In [75], it is mentioned that

TCP is only supporting reliable and strictly ordered data delivery. On the other hand, UDP just support nothing for reliable and ordered data delivery. There is no middle option of those. In addition, there are missing features in TCP and UDP such as multi-homing and TCP Friendliness.

SCTP [75, 76] also supports reliable transport which is more configurable than TCP. In addition, SCTP can support UDP-like features such as unreliable and unordered delivery with Partial Reliable SCTP (PR-SCTP) [88]. Most of all, it supports multiple streaming and multihoming in the transport protocol level. When a web browser is used, multiple TCP connections are used for different pictures and objects. This delivery could work with multiple TCP connections, but SCTP can handle this with one connection. In this case, SCTP enables saving overheads such as flow control, congestion control, and connection setup/teardown. Multihoming is the technique which increases connection reliability and network performance, using one or more IP addresses through one or more network interfaces [89]. SCTP is able to support this feature in the transport layer, so it provides transparency to applications such as FTP. SCTP mainly uses this feature in order to provide reliable service using redundant connections.

DCCP [77] is an unreliable transport protocol supporting congestion control [90]. DCCP also includes features for connection setup/teardown, Explicit Congestion Notification (ECN), congestion control and feature negotiation. There are two specific congestion control schemes: DCCP CCID-2 [91] which is a TCP-like approach (AIMD) and CCID-3 [92] which uses TCP-Friendly Rate Control (TFRC). There is also another candidate congestion algorithm, CCID-4, in order to support TFRC for small packets [93]. Since SCTP also supports TCP-like congestion control, DCCP CCID-2 looks very similar to PR-SCTP. However, PR-SCTP supports multiple streaming which includes reliable and unreliable traffic whereas DCCP only supports a single connection. There is also a standardisation approach for delivering Real-time data using Congestion Control Protocol (i.e. DCCP) [94].

## 2.6 Summary

As background technologies, networking technologies, context and metadata, data replication and transport protocols have been discussed in detail. Heterogeneous networks include not only wired technologies but also IEEE standards for wireless technologies. In this chapter, wireless technologies such as IEEE 802.11, 802.15 and 802.16 to support WMAN, WLAN and WPAN were presented. For wired technology, Ethernet as LAN and various broadband technologies were also introduced. For the involvement of data handling, metadata uses context for automated data annotation. The automatically-generated metadata can support the use of advanced features for data processing. As an example of data processing, data assessment may reflect user interests through the keywords from user. With these ideas, data replication can envisage performance and cost-oriented data replication approaches for the delivery and storage of sharing content and metadata among the users. Furthermore, UDP, TCP, RTP, SCTP and DCCP are also introduced as transport protocols which could be used for contents delivery. In the next chapter, related work will be discussed in terms of next generation networks, data replication approaches and multiple-source streaming approaches.

# Chapter 3

# Related Works

## 3.1 Introduction

This chapter discusses approaches similar to those proposed by Smart PIN; in terms of network connectivity, context-awareness and content delivery over network types. Actually, network connection issues related to Smart PIN is very similar to approaches for **Next Generation Network (NGN)** supporting heterogeneous connectivity technologies. However, those are usually focusing on issues at the network layer. As a context-aware system, Smart PIN concentrates on higher layer issues, especially relation to users. In this context, there is a short discussion of **person-centric context aware system** approaches.

Content delivery over networks could be categorised into non-realtime and realtime. In this thesis, realtime approaches considered are multimedia streaming such as broadcasting or on-demand service. On the other hand, among non-realtime approaches which have relatively weak time constraints for content delivery, data replication systems are presented which support file sharing. Finally, since Smart PIN uses a utility function in its user-centric approach, this chapter discuses related work on utility functions including a general introduction and some interesting applications.

## 3.2 Next Generation Networks

With the advent of the Internet and latest advances in personal computer and mobile phone related technologies, everyday life is always connected to the network. In addition, technology advances are applied to new devices and networks. Both aspects benefit the users. There are not only physical technology advances supporting higher bandwidth but also integration of existing networks and emergence of new services. These kinds of efforts are often labelled as Beyond 3G [95, 96] or Next Generation Networks (NGNs) [97] as current network technologies are named as third generation. As a common term, NGN will be majorly used in this thesis.

The most interesting aspect of NGNs is that **they commonly use Internet Protocol (IP)** as their network protocol. One other aspect they consider is **interconnecting heterogeneous networks** including the Internet, cellular networks and consumer electronics networks assuming that there are multiple interfaces on a device, for example a GSM mobile phone having WLAN for Unlicensed Mobile Access (UMA). Because of this, closed networks (i.e. cellular network, etc) have started to support connection to open networks (i.e. Internet). Another interesting characteristic is that every business player in NGN aims at **supporting multiple services on their network** such as voice, data and video delivery at the same time. **Mobility issues** are also one of the interests in NGN, including node identification and handover not only in homogeneous networks, but also in merged fixed and mobile networks. These four aspects could be the most common properties in current NGN activities.

The fairly narrow NGN definition from ITU-T describes it as a packet-based network which supports **Quality of Service (QoS) control** in transport, which is independent of service-related function and which supports generalised mobility [98, 97]. It uses TCP/IP as basic protocol. In ITU-T NGN, fixed and mobile network are involved together, for example Universal Mobile Telecommunication System (UMTS), Digital Subscriber Line (DSL), cable, etc. as shown in Fig 3.1. The integration with IP Multimedia Subsystem (IMS) provides a realistic vision for NGN to be implemented. This NGN extends the connection to home networks through Home Gateway (HG) [99]. In addition, the Digital Living Network Alliance (DLNA) [100] vision connects consumer electronics and other devices and enables the devices to be reached through HG. However, this is a network-centric approach and too strict for various approaches and services for current Internet and consumer electronics. Especially, there is lack of consideration of the users characteristics, their device properties and their content.

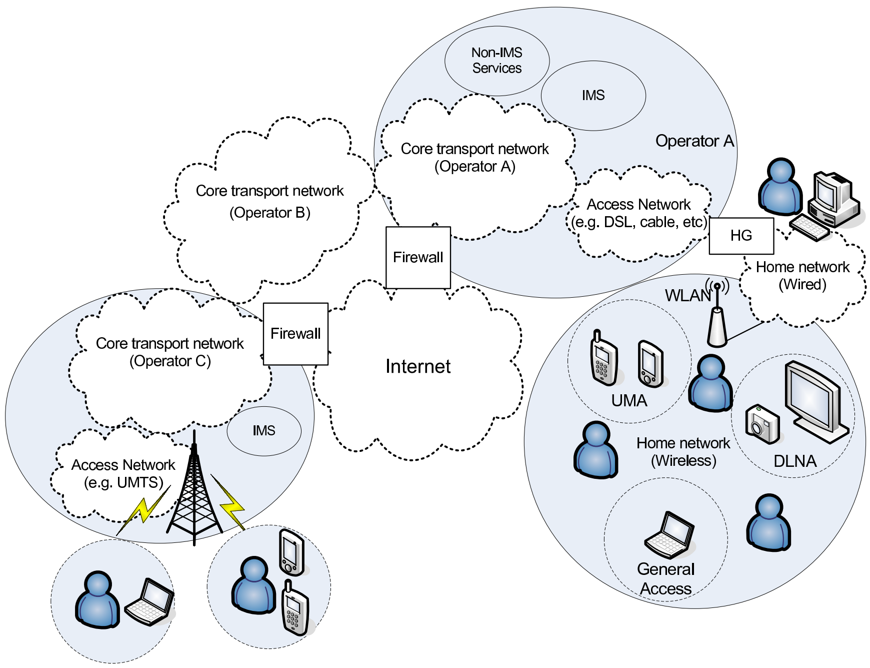



Figure 3.1: Simplified Architecture of Next Generation Network with IMS, HG and DLNA

Often, NGN is confused with Future Internet Initiatives (or New Generation Network [101]) which are activities aimed at building new network designs from scratch. In the US, GENI [1] is the Future Internet project. It also includes testbeds of Future Internet research which is similar to PlanetLab [2]. In Europe, Future Internet Assembly [3] deals with these issues as the FP7 EIFFEL project. In Asia, there are several countries which have their own projects. In addition, members of Asia Future Internet [4] cooperate with each other on this

[1] Global Environment for Network Innovations, http://www.geni.net/, last accessed 18 Nov. 2009
[2] PlanetLab, http://www.planet-lab.org/, last accessed 18 Nov. 2009
[3] Future Internet Europe, http://future-internet.eu/, last accessed 18 Nov. 2009
[4] Asia Future Internet, http://www.asiafi.net/, last accessed 18 Nov. 2009

topic.

Based on these clarifications, the following subsections discuss on the approaches in NGN as a broad definition. In terms of heterogeneous network support, they deal with how to use multiple network technologies, how to merge open- and closed-networks and how to support various services including voice, data and video in the area of Internet, cellular network and consumer electronics, respectively. At the end, the summary includes a short comparison with Smart PIN.

### 3.2.1 Internet-based Approaches

The Internet was originally designed to support best effort data transfer and utilises packets for transferring data using the Internet Protocol (IP) over wired network technology such as Ethernet. There are wireless networks such as Wireless Local Area Network (WLAN), Wireless Personal Area Network (WPAN) and Wireless Metropolitan Network (WMAN) and IP is still adopted as the mandatory protocol for the network layer. There are many efforts put in the interaction of wireless and wired technologies. Next, several systems which uses different approaches will be presented.

Personal networks (PNs) [96] refer to the extension of the personal area network to enable co-operation with remote devices and services. Since PNs are based on ad-hoc wireless networks, the system supports resource and environment discovery, self-organisation, routing, co-operation with fixed infrastructures, security, privacy, and accounting as its main features. Furthermore, Fednet was proposed as an ad hoc federation of independent PNs. A federation in Fednet could be considered as a trust-based access control scheme for shared resources. Based on this concept, My personal Adaptive Global NET (MAGNET) [5] focuses on network layer issues. In terms of service, existing applications are supported and there is no discussion about novel approaches.

Ambient network (AN) [102] is another approach based on wireless and wired Internet and could be considered as a control plane framework focusing on an automatic roaming

[5] MAGNET, http://www.telecom.ntua.gr/magnet/index.html, last accessed 18 Nov. 2009

agreement and composition of networks. For these features, AN utilises an AN interface (ANI) and builds an AN Control Space (ACS) using a Generic AN Signalling protocol (GANS). In AN, there are three Functional Areas (FAs): Composition FA, Mobility FA and QoS FA in order to support these mentioned features. AN focuses on self organisation from the view of the service provider. As AN includes new network designs from other networks, it requires higher deployment cost than other approaches.

Open Broadband Access Network (OBAN)[6][7] aims at allowing private or business broadband access to be shared with wireless vicinity users. As a result, users can save related costs using cheaper wireless facilities such as WLAN instead of expensive ones such as GSM. Thor-Gunnar Eskedal et al. [95] discussed that WMAN and WPAN could also be integrated and utilise some modules in GSM/UMTS for profile and policy/security information. This approach does not consider any improvement of the application, though.

A wireless mesh network (WMN) [103] could be considered as one approach using both wireless and wired technologies since it is also assumed that ad-hoc wireless networking supports connections between mesh routers. Some of mesh routers have gateway support, and they provide the other routers with connections to the Internet. The major difference from wireless ad-hoc networks is that mesh clients just have a connection with mesh routers instead of being involved in routing. In addition, the mesh routers are rather static in terms of mobility. WMN is a relatively good solution in terms of network deployment. However, once again there is a lack of consideration of application for user, content, etc.

### 3.2.2 Approaches in Cellular Networks

Although cellular networks were built to support voice services and high user mobility, data services such as text messages and multimedia streaming have become increasingly more popular among users and have become important revenue sources for operators. However, these services are based on their own networks and do not allow to connect to the server

[6] Open Broadband Access Network (OBAN), http://oban.prz.tu-berlin.de/, last accessed 25 Jul. 2007

[7] OBAN - Open Broadband Access Netowrk -SINTEF, http://www.sintef.no/Home/Information-and-Communication-Technology-ICT/Communication-Systems/Projects/OBAN---Open-Broadband-Access-Network/, last accessed 18 Nov. 2009

which is located outside of the cellular network, a so-called closed-network. In order to provide rich services offering true mobile Internet, it is necessary to support connection with other networks and via them to external servers. In this context, several approaches in cellular networks to support heterogeneous connectivity are presented.

The Fixed-Mobile Convergence Alliance (FMCA)[8] is a global alliance of operators to develop convergence devices supporting WLAN and GSM through Unlicensed Mobile Access (UMA). This looks very similar to the OBAN project (presented in section 3.2.1) but the stance of the application is different. In other words, FMCA focuses on mobile phone applications such as voice call. However, OBAN is based on WLAN which aims at data applications. In addition, OBAN is extendable to other technologies such as WMAN and WPAN.

Virtual Home Environment (VHE) [104] is a concept based on UMTS for Personal Service Environment (PSE) portability across network boundaries and between terminals. The concept of VHE is that users are consistently presented with the same personalised features, user interface customisation and services in whatever network and whatever terminal (within the capabilities of the terminal and the network), wherever the user may be located. VHE supplies a toolkit for presenting services that a user subscribes to, a device can support and a network can provide. However, this approach is too ideal to abstract all new services and to support quick changes of its applications.

The 3GPP[9] standards include IP Multimedia Subsystem (IMS) [105], proposed for evolving steps for 3G cellular network in order to support delivery of IP multimedia services to the user. Based on the current GSM/GPRS and UMTS networks which are mixture of circuit-based and packet-based networks, the standards add on features of IMS utilising some IETF specifications such as Session Initiation Protocol (SIP) as 3GPP network is evolving to an all-IP network. With this SIP support and network advances, existing services (i.e. Voice service, Dual Tone Multi Frequency (DTMF) tones, etc) and new services (i.e. presence information, instant messaging, etc) are provided through cellular networks

[8]Fixed-Mobile Convergence Alliance (FMCA), http://www.thefmca.com/, last accessed 18 Nov. 2009

[9]3rd Generation Partnership Project (3GPP), http://www.3gpp.org/, last accessed 18 Nov. 2009

or other broadband networks (DSL, cable, etc) [98]. Although heterogeneous networks are supported, this approach focuses on network and services instead of the users and their content.

### 3.2.3 Consumer Networking Approaches

This category of approaches mostly involve consumer electronics, and assume that WLAN or WPAN are used for interconnecting devices and/or enabling devices to connect to other networks. Specifically, the Digital Living Network Alliance (DLNA) [100] is a most popular standard for consumer electronics to enable the interaction with PCs and mobile devices. The main service of DLNA is seamless digital media and content sharing. DLNA services are operated over the home network, and include scenarios supporting some mobility. DLNA-enabled devices uses Universal Plug and Play (UPnP)[10] which is a group of protocols offering the basic features of DLNA over IP. Mainly, UPnP defines Simple Service Discovery Protocol (SSDP) delivered through UDP, service control through Simple Object Access Protocol (SOAP), and event notification of service through Generic Event Notification Architecture (GENA). In addition, they use eXtended Markup Language (XML) for service and data description.

In this context, this section introduces several UPnP-based approaches [106, 107, 108]. DLNA and UPnP standards cover devices within a home network. In addition, data sharing is based on a server-client architecture. These approaches consider heterogeneity of the available connectivity technologies including wireless and wired networks and different characteristics of data to be distributed over the networks as mentioned in section 1.2. However, most approaches do not consider user's interest and quality of multimedia delivery over the network.

SHARE [106] located in the Home Gateway (HG) supports content sharing among UPnP-enabled home networks. SHARE supports to share content UPnP Audio Video (UPnP AV) components (e.g. MediaRender, Control Pointer and Media Server), and pro-

[10] UPnP Forum, http://www.upnp.org/, last accessed 18 Nov. 2009

vides content search and streaming using three additional modules such as HomeConnector, MediaDistributor, and VirtualMediaServer. Although SHARE proposes RTP-based multiple source streaming, its focus is only the compatibility with UPnP standard and does not consider performance and cost-related aspects in relation to the users and their contents.

Wide-Area Media Sharing (WAMS) [107] is an approach supporting content sharing across wide-area networks based on DLNA/UPnP. Since UPnP does not support group management, WAMS use overlay middleware which deals with group creation and maintenance. In addition, they use extended UPnP (xUPnP) due to lack of support for wide-area networks. xUPnP simply collects UPnP local device information and relays to remote xUPnP devices which will forward this information to its local UPnP devices. Although WANS enables the user to connect her devices which are widely distributed in different networks, there is no consideration of performance and cost of content access and delivery.

Intelligent Home Environment (IHE) [108] is also a UPnP-based approach to support P2P networking. In order to achieve this, they also employed middleware between UPnP and the underlying TCP/IP protocol. However, UPnP messages are delivered through P2P overlay networks instead of using its protocol directly. They also proposed a service collaboration framework based on XML. The framework supports how to plan that the services collaborate in order to reduce unnecessary user involvement. This approach focuses on how to implement P2P overlay network establishment and its maintenance but it does not deal with cost efficiency during delivery of content.

There is also an approach which is not actually based on UPnP. Ubiquitous peer-to-peer (UP2P) [109] is a P2P overlay network service which is based on two-tier variable hop overlay, Chameleon [110]. For interoperability between other networks, UP2P use a hybrid infrastructure and ad hoc network for wireless and overlay network federation. Instead of using Session Initiation Protocol (SIP), the authors proposed P2P SIP [111] using Dynamic Hash Table (DHT) which stores user location information. One interesting aspect is that the proposed solution supports overlay multicasting for multimedia delivery [112]. UP2P supports not only best-effort delivery but also real-time streaming. Although UP2P uses DHT, recursive blind search [113] is adopted in order to enhance searching without identification for DHT entries. Although the authors consider data delivery over the UP2P, there is no consideration of cost which is related to content delivery.

Home Gateway Initiative (HGI)[11] is a nonprofit organisation focusing on home gateways used to connect devices in the home to the Internet or other WAN [99]. HGI defines six use cases such as broadband connection installation and service add-on, communication service similar to telephone, fixed-mobile and service convergence, home office, entertainment and information including IPTV, IP radio, gaming and Media Server, remote access for home devices as well as home management and security. Specifically, remote access includes control of both DLNA UPnP based devices and non-UPnP devices. These standards include server-client services and do not include other services such as P2P-based for connecting remote devices and home network devices.

### 3.2.4 Summary

In this section, NGN approaches in various areas were discussed. They support network heterogeneity as devices use multiple network technologies for their specific applications based on voice, data and/or video. In order to achieve high mobility and quality of services for various applications, the approaches are introduced using a categorisation based on the research areas which are Internet-based, cellular and mobile networking and consumer networking. In case of Internet-based approaches, there are various and flexible solutions but only some of them such as WMAN, WPAN, etc. are considered in order to support network heterogeneity and device mobility. As cellular networks are evolving towards all-IP networks, IP Multimedia System (IMS) focuses on supporting different technologies such as UMTS and broadband connectivity supporting quality of service at transport level. Consumer networking approaches use UPnP over IP protocol and several approaches enable remote devices to communicate with other devices behind Home Gateways (HG) in a P2P manner. However, most solutions only focus on network issues and limited services and

[11] Home Gateway Initiative, http://www.homegatewayinitiative.org/, last accessed 18 Nov. 2009

applications without consideration of quality of service.

## 3.3 Person-Centric Context-Aware Systems

In order to make a system smart, context-awareness is an essential feature and metadata is a good container for context related to contents, as discussed in the previous chapter. Usually a context-aware system detects a user as normal context information. For example, the system distinguishes and acquires which user has interaction with the system. However, person-centric approaches handle user information differently. Person information is chosen as central idea among context and other related context is coupled when it is related to it. As an example, a user can take pictures during his visit to Dublin for a meeting. In this case, context information such as location and occasion can be associated with the user, and this information could be used as metadata for the pictures. As person-centric context could be used for user's content as metadata, it could be easy to implement a user-centric system with this approach, too. Three systems will be discussed next as examples of person-centric approaches in context-aware systems.

Context Shadow (CS) [42] provides a way to create searchable clusters of context information related and useful to a user. CS adopts a blackboard based and person-centric architecture for gathering context data. The blackboard approach enables an efficient interface for sharing context data between devices. CS assigns a context server to a person's context and enable application interfaces to access that. With this assumption, services are able to work on a more abstract level. As discussed, CS focuses on the context modelling issue and does not address delivery performance or cost issues.

Personal System (PS) [44] is a second example of a person-centric context-aware system. In this case, the overall system consists of sensors which are located very close to the user in a WPAN which send context data to a centralised system. PS has two parts: the context information system (CIS) which collects and manages various information related to a user from different sources, and the application system which uses that context information from the CIS. For the unreliable data connection, PS uses communication metadata which enables it to manage connectivity information and to use it again when possible. PS also supports location sensing with Cell-ID which is matched with location information stored in a database. Furthermore, PS includes the Anonymiser Proxy Server which enhances user privacy during communication with untrusted parties. In spite of its benefits, PS does not address data delivery performance or cost-related issues, either.

A third solution is OmniStore [63, 64] which adopts a semantic file system based on archival storage. OmniStore proposes beacon-based dissemination of context and automatic context annotation of files. OmniStore uses an architecture which includes personalised devices in WPAN. This is also adopted as one of the basic network topologies in Smart PIN. OmniStore focuses on detailed issues of dissemination of context such as propagation and control. It also deals with automatic annotations, applying context to data which is generated in each device using an annotation handler. Even though this approach considers some aspects of uncertain characteristics of WPAN, the authors do not focus on those issues in detail. They do not consider performance or cost issues, either.

Most of the person-centric context-aware systems do not concentrate on network-related issues since they mostly exchange small size context data. As context annotation is getting popular and has various applications, more and more context metadata will be shared among devices. Also, content is increasingly exchanged, and puts pressure on the networks. Consequently, context-aware systems should also deal with performance and cost in relation to content delivery as Smart PIN does.

## 3.4 Data Replication Systems

Data replication systems are distributed data processing systems to enhance performance and reliability using intentional duplication of its data among the member devices[114]. As depicted in Fig 3.2, the literature for data replication mostly focuses on consistency issues in relation to data exchanges [66, 65, 67, 63, 115, 116]. Some data replication systems consider data availability issues for managing their different replication costs, and some focus on fault-tolerance issues for providing proper data in spite of failure of some devices [60, 117, 118]. Since P2P systems embed fault-tolerance characteristics, discussions on consistency management and data availability issues will be presented here. At the end of this section, data replication algorithms will also be introduced.

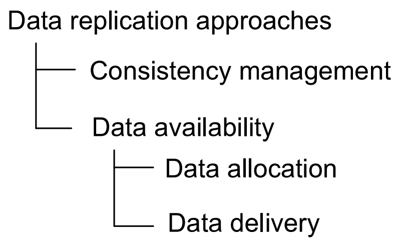



Figure 3.2: A data replication approach categorisation

### 3.4.1 Consistency Management Approaches

One of the main issues from previous research on data replication is the consistency problem which is caused by access to distributed data. Most systems assume that they are cooperative or group-based. With this situation, since there could be asynchronous read and write access of data, they concentrate on how those operations could be managed to keep the consistency of a file. There are two approaches: read-any-write-any (RAWA) and write-once-read-many (WORM).

RAWA allows for users to modify a file anytime they want. Due to this, there is a problem of consistency which needs to be resolved or prevented. For the RAWA replication scheme, there are two types of approaches: an optimistic (asynchronous) scheme and a pessimistic (conservative or synchronous) scheme. For the optimistic scheme, there should be update synchronisation and a conflict resolution algorithm for each architecture [66, 65, 67, 119]. In the case of the pessimistic approach, there is only one replica that could be changed at a time. To make this work, usually one original piece of data is kept among the multiple replicas [120].

For simplification of the consistency problem, a WORM model has been used that create a new version every time when there is a write. This approach is used in OmniStore [63, 64] and Cedar [115, 116]. However, this approach has a side effect such that there might be too much similar content if there is only small change in different versions of contents. This issue could be small for small-sized files but big-sized data such as video could generate unacceptable results in the system. The detection techniques for near-duplicated content [121, 122] could be useful for this situation.

### 3.4.2 Data Availability

Another issue addressed in data replication system is data availability or durability. Data availability means the ratio of successful data accesses over total requests. If data is not reachable from a mobile device, the access is considered as a failure. Related to this issue, data allocation and delivery are the main aspects to be considered.

#### 3.4.2.1 Data Allocation Approaches

The approaches for data allocation involve the **selection of data which will be replicated and the choice of devices which will store the data**. The simplest way could be an all manual approach [67]. Users can select the data and copy that to the device they want. A simple automatic way could randomly select data and location in order to generate duplication [65]. In a smaller system, it might be better to consider an efficient way to access data successfully [123]. The discussions for each approach will visit their details.

**Roam** [67] is a hybrid approach between central and distributed data replication systems, and provides P2P-based optimistic replication assuming that there is a mobile situation in which users leave and have no available network connection. It defines a wide area replication domain (WARD) which manages a unified set of replica sets in mobile devices. Since there is no specific server or original copy of the data, a node might have different replicas for each other. Furthermore, the system can support dynamic generation and deletion of data keeping a total set of files in a P2P system. In this context, Roam proposes a selective replication scheme for their system [124]. The two ideas in this scheme are status vector and full backstoring. The status vector stores the *file id* and the *visibility of files* which is in a local file system. This data is independently and optimistically managed. Full backstoring is the way to include the parent directory of locally stored data. If a file is stored locally, then its parent directory also will be kept in the local file system. As shown, there is no algorithm for autonomous data and device selection for replication, leaving the user responsible for data selection on the device for future use.

**OceanStore** [65] is a large scale, networked data replication system based on servers of unpredictable operability to provide nomadic data accessibility. The goal of the system is global-scale persistent storage working on an untrusted infrastructure including servers having variable fault ratio and handling nomadic data available anywhere, anytime. This system handles data as changing active objects and read-only deep archival objects. For the data allocation issue, random data allocation for load distribution is used. It provides a unique id for each server and distributes data using the server id as a portion of data id. Specifically, OceanStore uses a fixed number of erasure coded fragments for deep archival data. The erasure coded fragments are small pieces of data which have redundant codes for forward error correction. If these erasure coded fragments are gathered, the original data could be reproduced. However, the devices considered are not mobile devices and the system does not deal with characteristics of different network technologies.

**Wayfinder** [70] is a P2P file system supporting optimistic consistency management based on PlanetP [125], a toolkit for medium-scale P2P applications. PlanetP includes three sub-systems: a gossiping module, an index storage system and distributed query processing engine, and a lightweight active distributed hash table. The design goal of Wayfinder is to provide a global namespace as a shared view of the file system instead of mounting each device into a specific directory, and content-based queries accessing data through not only browsing with file name but also searching with content properties. Furthermore, Cuenca-Acuna et al. [123] propose **autonomous replication (AR)** to achieve high availability of data in P2P file systems based on PlanetP. In this, the erasure coded fragments are replicated randomly on the free space of peers. The estimated availability for a file and fragment is measured periodically. As mentioned, Wayfinder supports middle-scale data systems. If a node which has a lot of new data joins, the system needs to spend quite a long time to achieve target data availability. Furthermore, Wayfinder does not focus on heterogeneous networks.

#### 3.4.2.2 Data Delivery Approaches

The approaches for data delivery involve scheduling of data delivery for replication among the devices. These could be categorised as reactive and proactive approaches. The reactive approach initiates delivery when it is required. Instead, the proactive way initiates delivery before it is required. This section discusses delayed delivery as a reactive approach and periodic and budget-based delivery as proactive approaches.

**Delayed delivery** manages a queue of delivery requests and waits until the connectivity between devices is available. OmniStore [63, 64] adopts schemes which are similar to a computer's cache memory such as file off-loading, push caching and backup schemes. Specifically, push caching used for transferring files to mobile devices from repositories, and a backup scheme used for replicating files from mobile device to repositories, are performed on this basis. Even though OmniStore considers various connections such as WPAN and Internet, there is no consideration for different costs of each connection.

**Periodic delivery** works with some conditions for determining whether there is any need to transfer data. The basic scheme for delivery used in OceanStore [65] is a periodic approach for erasure coded fragment distribution using a very low rate such as once a month. However, this is just for repairing the fragments which are considered as deep archival files. The automated replication scheme [123] based on Planet P [125] also uses the erasure coded fragments. The estimated availability for a file is measured periodically. Nodes of the systems share this measured data availability. If the availability of a file is not reached to target availability, a randomly-generated fragment of file is distributed to a randomly chosen node. Since the system randomly chooses data to transfer, this approach does not consider how important data is to the user.

In **Tempo** [126], a proactive method of replication of data during idle time of devices is proposed. To limit the usage of bandwidth, it introduces a bandwidth budget, which defines the maximum data size per unit time. With this user-specified parameter bandwidth budget, it removes bursty data transfer for repairing data fragments in a reactive way and provides the same level of durability as the previous implementation, with no fluctuation of data transfer. The decision to transfer is also defined using the bandwidth budget parameter. This approach does not consider user interest in data though.

As discussed, data delivery in data replication could be reactive or proactive. Although there is significant consideration of performance and cost parameters, there is a paucity of user's interest in data in previous research. If there is no consideration of user's interest, expected satisfaction could vary with every situation user has.

### 3.4.3 Data Replication Schemes

In addition to the discussions above, an issue that needs to be addressed is how data replication works. Fundamentally, the schemes could be categorised into manual and autonomous approaches. In **manual schemes**, the selection of data, location and delivery is dependent on user activities as Roam does [67]. In this system, users pick up the data needed to be carried and stored on their devices. In contrast, the system adopting an **autonomous data replication scheme** performs that process by itself. The user just needs to set up the parameters or configuration, explicitly or implicitly. This is the preferred approach for future systems.

If there is too much data to handle, autonomous data replication should be involved. An autonomous data replication scheme could be classified into static or dynamic schemes [127]. The **static replication scheme** indicates that data replication is determined at system deployment time and only changed with administrative control. In other words, no adaptive data replication properties are provided during system operation. Even though this static scheme does not consider dynamic characteristics of traffic and networks, this scheme is similar to the static file assignment problem which is the NP-complete. On the other hand, the **dynamic replication scheme** measures an objective metric for network, user behaviour and other performance parameters and provides adaptation to achieve specific requirements. Compared to the static scheme, this requires continuous monitoring of performance parameters.

As addressed, the data replication scheme should be autonomous if user involvement for data management is difficult. Specifically, data overload require an autonomous approach for data replication providing user satisfaction with adaptation of performance and cost of the system. In next section, a discussion on the algorithms for dynamic data replication especially for a mobile environments is presented.

#### 3.4.3.1 Algorithms for Autonomous Data Replication

In traditional Internet data replication systems for the Web, various algorithms such as knapsack, bin packing, capacity-constrained optimisation, game theoretic approach, etc. are used [128]. In the application area of data replication, many of these algorithms are used to solve specific problems such as for example game theory and the knapsack algorithm [129].

For data replication in a mobile environment, **Geels and Kubiatowiczthis** [130] discuss the necessity of an **economic model** for data allocation. They categorise replica management systems into Content Distribution Network (CDN), P2P and Economic Model approaches. Especially, they present benefits of the economic model and their directions of research. To compare with the centralised model, a distributed selfish economic model looks complex and difficult to achieve maximum level of performance. The economic model assumes that each node works independently, and consequently provides higher scalability than other approaches. For this reason, the economic model can provide a reasonable solution for low computing power devices in networks, especially in large systems.

**Chun et al.** [61] adopt a **game-theoretic approach** which could be considered as an economic model for file replication. They propose game models such as a basic game model and a payment model, and show that the models have Nash equilibria which is considered as optimal solutions of game theoretic models. Additionally the researchers simulate how efficient these models are. They define the payoff (actually cost of the game) as summation of placement costs for data which is replicated into the server and products of demand and distance for the objects which are not replicated in the server. For the payment game, they add bid and payment factors into the cost function. Furthermore, the social cost is defined and targeted to be minimised. The test is based on the Mosek simulator[12] for solving integer linear program to get the social optimum configuration. The test involves several topologies such as line topology, transit-stub topology and power-law topology. However, this scheme assumes that only the servers which get the request do the replication, but clients do not. In mobile systems, the mobile devices need to be separate peers and to use replication, too.

Another interesting approach is using a **colouring algorithm with graphs** [131]. Under the resource allocation problem, the Asynchronous Distributed Colouring (ADC) protocol is proposed. Each node tries to put different resources in its neighbourhood which it does not have, and it also tries to put the same resource as far as possible from itself. Simply, the resource could be considered as different colour and the algorithm translate into the allocation of that colour on each node measuring each distance to identical and different colours.

**Ranganathan et al.** [132] propose an **autonomous dynamic data replication scheme** for unstable, decentralised P2P networks. First of all, the number of replicas is calculated with node availability, replica location service accuracy and file availability. In this model, in proactive mode the node checks periodically, and in reactive mode it checks when a file is not accessible, in order to determine when data should be replicated. When it is required to be replicated, the node chooses a proper node to be replicated. For this, cost of transfer in time is calculated and the benefit of replication is estimated based on the unit of transfer time as well. With these parameters, the most suitable node can be selected.

[12]Mosek, http://www.mosek.com/, last accessed 18 Nov. 2009

### 3.4.4 Summary

In this section, detailed issues and approaches for data replication are presented. Consistency management, data availability and data replication schemes are the main points. Smart PIN also considers these issues and provides a novel approach based on performance and cost-oriented utility based data replication. This approach uses metadata for data, performance and cost metrics for network and system for decision of data selection, allocation, and delivery. Especially, Smart PIN helps handle large amounts of data and reflects users' interests during system operation.

## 3.5 Multimedia Streaming Approaches

### 3.5.1 Overview

Smart PIN supports various types of data including multimedia. This multimedia data can be delivered in a non-realtime or realtime manner. In a realtime manner, multimedia data should be delivered in the time given as a time constraint. When multimedia is delivered within these constraints, a different service could be possible, for example, a broadcasting service supporting live sports events and on-demand services supporting movies through the network.

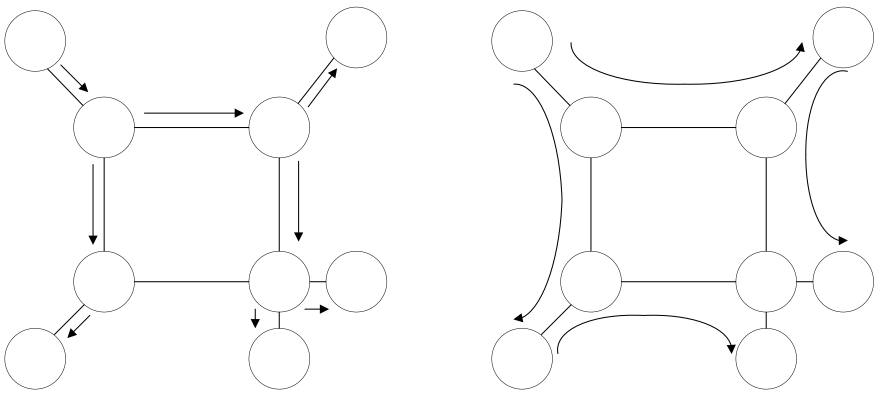
During the service, clients may access one server or multiple servers depending on the scheme they use. Recently, research has presented various benefits of multi-source streaming compared to single source streaming. In addition, quality of service (QoS) of streaming is also very important in order to provide better user perceived quality although there are several challenges such as network condition variance, etc.

This section discusses how streaming approaches support broadcasting services and on-demand services. In addition, comparison of single- and multiple-source streaming approaches is presented with benefits and overheads. And finally, different approaches for achieving high QoS levels are also introduced.

### 3.5.2 Service based Categorisation

#### 3.5.2.1 Broadcast Streaming

**Broadcast streaming could be defined as streaming live multimedia** through a network. Single services such as TV broadcasting could be simply supported using broadcasting solutions. However, interactive services may also need to be supported, and then multicast could be the best solution for broadcast streaming since it supports point-to-group data delivery. Multicasting approaches could be categorised as **IP multicast** [133, 134] and **Overlay multicast** (or Application level multicast, ALM) [135] as shown in Fig 3.3.

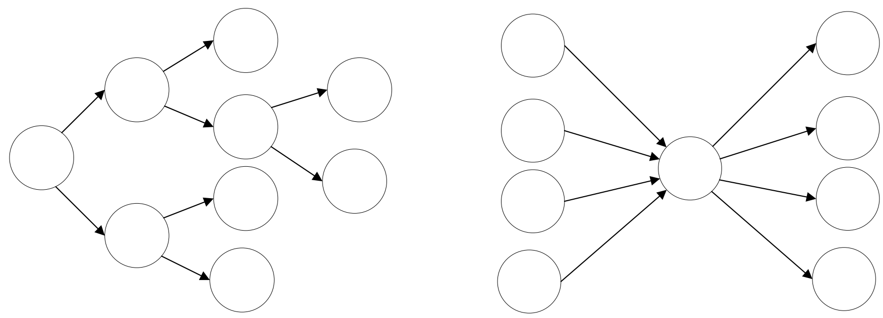


(a) IP multicast (b) Application layer multicast

Figure 3.3: Multicasting approaches

**IP multicast** [133, 134] uses a specific class of addresses for indicating groups for multicast. In the view of the end-to-end, a sender just needs to send data to a specific IP address and a receiver also receives data from a given IP multicast address. However, the mechanism to distribute data is mostly deployed on routers which support packet routing protocols. There are additional protocols dealing with mechanisms for creating and joining groups. The biggest drawback with IP multicast is deploying cost. Since the transport layer is involved, the application just needs to focus on other issues. However, most of the routers in the Internet should be changed and this could be considered as one of the reasons that makes IP multicast deployment slow. Recently, it is still considered for delivering multimedia data such as IPTV broadcasting services because it is efficient from the view of service providers [136].

As shown in Fig 3.4, **overlay multicast** could be categorised based on the architecture of overlay in **tree-based** [137, 138, 139, 140, 141] and **mesh-based** [142, 143, 144, 145]. Tree-based approaches are actually part of single-source streaming since each receiver has only one sender in the tree structure. Mesh-based approaches use connections between multiple senders and multiple receivers. These different structures provide various trade-offs in terms of communication efficiency, robustness, scalability, maintenance costs, etc.



(a) Tree structure (b) Mesh structure

Figure 3.4: Application layer multicast categorisation

Among tree-based approaches, **Cooperative networking** (CoopNet) [137] is a tree-based streaming approach based on data caching, storing the delivered data for future use. It adopts Multiple-Description Coding (MDC) to provide a robust service against disturbance caused from frequent join and departure of clients, maintaining multiple distribution trees.

**Zigzag** [138] provides an administrative organisation algorithm which has bounded numbers of children and the tree depth is set to the logarithm of the number of nodes. In order to achieve these characteristics, it was proposed to build clusters which include peers. Based on these clusters, a head and associated heads are selected. These heads of a cluster take the role of servers to peers in lower levels of the cluster. The connection between a server and clients is built using overlay multicast tree structure construction scheme. The overlay multicast tree structure is separated from administrative organisation. Because of this, the approach manages two data structures for streaming, and it provides enhancement of the performance.

**SpreadIt** [139, 140] is also a tree-based approach. When a new receiver joins to live streaming, it searches for a source which has room for sending data to the receiver. The authors define protocols handling join and departure of peers and mechanisms maintaining the tree structure which delivers multimedia to peers. Especially, a Smart Placement scheme is employed when a new peer joins the streaming that utilises trace routes for finding the best position which a new peer connects into the tree. The authors also call the approach PeerCast.

**SplitStream** [141] supports two main features, namely content distribution and multimedia streaming. In SplitStream, contents are segmented into stripes and distributed to participating peers in Scribe [146], which is a ALM system. Scribe uses a DHT, Pastry [147] which is the overlay of content distribution in order to support searching the contents. The strip could be generated through erasure coding or MDC. These split contents are multicasted in a separate application level multicast tree.

**As a mesh-based approach**, **Data Overlay Network (DONet)** [142] is the first approach for a gossip-based P2P live streaming approach. DONet nodes cooperate with their partner: exchanging a Buffer Map, describing what data they have and delivering required packets using a transport protocol such as TFRC. In addition, they use a deadline-oriented greedy algorithm for assigning packet delivery from multiple sources.

**PROMISE** [143] is a P2P multimedia streaming system which supports peer selection and multiple-source streaming based on CollectCast, P2P services which support collecting network performance and monitoring the status of peers. Two approaches are proposed for peer selection based on estimating the network, namely topology-aware and end-to-end peer selections. End-to-end peer selection uses available end-to-end bandwidth. Topology-aware peer selection is based on network tomography [148] which enables examination of network internal characteristics from end-to-end characteristics. Selected peers are used as active senders in order to stream to a receiver.

**Narada** [144] is a small scale multi-source streaming. It is also assumed to support multiple receivers. This approach targets replacing IP multicast with end system overlay (i.e. P2P systems) using a self-organising and self-improving approach. Instead of that, it can save deployment costs for IP multicast. However, it is based on a mesh architecture which may generate more overhead to maintain the streaming structure.

**AnySee** [145] could be mentioned as mesh-based approach. However, multimedia data delivery is performed through an overlay tree architecture. To compare with other approaches, AnySee adopts an inter-overlay optimisation scheme to resolve issues such as long start up delay, end-to-end delay and low resource utilisation. Usually, these problems are challenged within the given delivery overlay tree. In contrast with this general intra-overlay optimisation, peers which are located in different overlay trees are used, as well.

#### 3.5.2.2 On-demand Streaming

**On-demand streaming assumes that there is stored multimedia content and provides streaming based on user requests**. Traditionally, this service is usually based on a server which provides content using unicast delivery channels for each client. However, there are also approaches using the multicast concept to reduce overhead for extremely popular content, etc. In addition, there are also approaches to distribute the load of a single server with P2P-based concepts. In this section, **multicast based single server on-demand approaches** are mentioned first. After that, **distributed server based schemes** are presented.

The Group-based Multimedia on Demand (GMoD) approach uses **IP multicast** for efficient use of resources such as the channel. Huang et. al [149] adopted a batching method which combines several nodes requesting a movies around same time and provides them with the same multicast channel. For the wireless environment, the authors derived a new analytic model based on a performance metric such as blocking ratio, and tested that model via simulations. Based on simulation results, an optimal timeout-based scheme was proposed and was compared with existing algorithms. For the analytic model, the metrics for performance were satisfaction ratio, reneging ratio and blocking ratio. Also, the researchers showed that the new scheme has better performance from the view of the metrics they introduced. As it is based on multicast, this approach needs to have a group sharing a multicast channel, but there is no cooperation between receiving nodes.

It is very popular to use **Overlay multicast** approaches not only for broadcasting services, but also for on-demand services, since it lowers the load of the server. As most overlay multicast approaches are **P2P approaches** (i.e. ALM), they will be described next.

**P2Cast** [150, 151] adopts a patching technique in P2P streaming. Basically, clients receive data through a tree which is built for ALM. In addition, a receiver gets a patch from other nodes which are not their parents. For example, a newly added receiver (i.e. node A) uses its session to a parent (or server) and also listens to a receiver (i.e. node B) which is not a parent (or server).

**P2VoD** [152] also uses patching which a peer gets from others. The difference is that the P2VoD client caches the most recent content while P2Cast [150, 151] caches the initial part of the content. In addition, P2VoD only uses one relay stream. Because of this, P2VoD manages generations which are the groups of peers having the same data in their cache.

**oStream** [153] provides VoD services, based on an application level asynchronous streaming multicast mechanism, introducing cache and relay. The solution adopts a buffer for storing data which was already played. If there is a newly joined node, it gets data from a node which caches required data. The cache is implemented as a fixed length circular queue.

**dPAM** [154] introduces the prefetch and relay scheme instead of the cache and relay scheme [153]. In contrast to cache, the prefetch buffer uses a lookahead scheme storing data which will be played. The assumption they have is that there is excessive bandwidth to receive data which is currently playing. In addition, the scheme uses memory to save data which the node will play later. The stored data is relayed if there is enough bandwidth.

These ALM based approaches provide load distribution from the server, but there should be a server which contains all the content. In contrast to that, several approaches which use multiple servers containing the whole or part of the content [155, 143, 156, 85], envisage on-demand features using **mesh-based approaches**.

As an example, **VMesh** [155] proposes an architecture for interactive VoD services in

P2P networks based on Distributed Hash Table (DHT) such as Chord. This DHT provides the receiver to find the wanted segment efficiently and benefit from random jumps for a VoD service. It divides videos into variable length segments and stores them in distributed peers over the Internet. The authors adopt a locality-aware segment location algorithm, providing less stress to the server and good quality to the client. The popularity based segment storage scheme improves playback continuity. VMesh estimates the popularity of segments and determines which segments are kept stored and which are deleted in adaptive and periodic approaches.

However, existing approaches for Video on Demand (VoD) provide limited services in terms of supporting high QoS levels. In other words, most provide limited and equal quality, since all nodes share same multimedia data and those are usually encoded at constant bit rate. More discussion on QoS provision will follow in the last subsection of this section.

### 3.5.3 Single-source vs. Multi-source Streaming

Streaming is basically assuming a client-server architecture with a server serving several clients. **Single-source streaming approaches** involve one server streaming multimedia to any client. In ALM, most of the streaming is single-source. Tree-based ALM approaches [138, 152, 153, 154] are all single-source streaming, since each receiver usually has one server except some schemes using patching [150, 151], multi-tree [137], or Multi-Description Coding (MDC)[141].

Compared to single source streaming, **multiple-source based approaches** show quite good performance in order to resolve issues such as varying network conditions during streaming. In addition to some tree-based approaches [150, 151, 137, 141], the mesh-based ALM solutions are usually using multiple sources [142, 143, 157]. However, these approaches use multicast which does not support high QoS provisioning which is not desired [158].

There are also several approaches [156, 85] which use unicast-based multiple source streaming. Compared to multicast-based approaches, unicast-based streaming has several strengths, mostly in terms of QoS support. Since the best-effort network characteristic generates a great hurdle for these approaches, multi-source usually provides a greater chance to overcome those limitations. There could be two kinds of approaches in terms of division and assembly of content for delivery approaches: **Interlaced Packet Assembly (IPA)** and **Multiple-Description Code (MDC)**.

**Nguyen and Zakhor** [85] proposed a framework for streaming video from multiple mirror sites simultaneously to a single receiver on the Internet. The scheme is based on a receiver-driven protocol which targets achieving higher throughput and increased tolerance to loss and delay due to network congestion adopting a rate allocation algorithm (RAA) and a packet partition algorithm (PPA). PPA supports interlaced multimedia data delivery from multiple sources for the receiver. They use IPA from multiple sources for streaming.

Among solutions using MDC, there are P2P Adaptive Layered Streaming (PALS) [156] and CoopNet [137]. Especially, PALS is a receiver-driven approach based on the adaptive delivery of stored layer encoded streams from multiple sender peers to a single receiver. It is built on its own quality adaptation for congestion controlled playback of layer encoded video over the Internet.

In multiple-source streaming, IPA benefits from efficient usage of network resources reducing replicated transmission of data with reasonable overhead. However, it does not look differently after the data which is more important to users than other data. MDC approaches are very good for lossy environments, but there is too much overhead to deliver duplicated information which is considered important.

### 3.5.4 QoS Handling in Streaming

QoS could be defined as a set of characteristics of a multimedia system to achieve the required functionality of an application [159]. One of the issues to achieve QoS of streaming could be mismatching of available network bandwidth and media sending rates. There are two ways to solve this issue: **rate adaptation** [86, 87] and **buffering** [160, 161]. The rate matching approaches solve the lack of network resources problem by adjusting multimedia quality. The approaches with play buffer provide more flexibility to the application but suffer from unexpected stops in playing caused by buffer underflow. This causes additional delays which is not expected. Buffer underflow management solutions enable better QoS, in this context.

There are several **rate-adaptation approaches** enabling application and transport layers to share the rate of multimedia stream and estimated available network bandwidth [86, 87, 84]. The approaches could be categorised regarding how they handle the mismatch of bit rate of application and network. These are good for adapting to network conditions but sacrifice multimedia quality to compensate.

**Yan et al.** [86] proposed a Media- and TCP-Friendly Rate-based Congestion Control (MTFRCC) protocol with MPEG4 fine-granularity scalable (FGS) coding scheme [162]. They define a convex optimisation problem with network condition and Rate-Distortion relationship of MPEG-4 FGS. Similar to TFRC, optimal bandwidth is retrieved as a function of loss and constants, depending on the video source stream, and provides media-friendly data delivery results. In order to achieve TCP-friendliness, the authors modify the bandwidth function for the throughput equation which is used by TFRC.

**Cuetos and Ross** [87] proposed an adaptive streaming approach for stored video over a TCP-friendly connection. Based on MPEG-4 FGS, the authors use a base layer (BL) and enhancement layers (ELs). At any time, the server transmits all of BL and part of ELs. The researchers formulate a stochastic control problem with some timing assumptions, in order to determine how much of the EL will be transmitted. In order to get an optimal policy for transmission, the authors introduce conditions for no loss, bandwidth efficiency, and rate variability. Since user-perceived quality easily degrades with loss of BL, optimum transmission policy needs to guarantee no loss in BL. In addition, an optimal transmission policy will show better performance when it uses higher bandwidth, since adding EL increases quality. The heuristic algorithm also considers the variance of bandwidth among possible transmission policies.

In [84], Yan et al. proposed an enhanced rate control mechanism which adapts application rate to network condition. TFRC is friendly with network conditions but not with user-perceived quality of media which is delivered through the connection. The proposed approach uses a state transition diagram which includes three states: High Rate (HR), Low Rate (LR) and Application minimum rate Aware (AA) states. HR state indicates that the TFRC estimated rate is greater than or equal to the application's average high rate which is required to transmit the base layer and enhanced layer of video. LR state is used for an estimated rate which is lower than the average high rate. Furthermore, this state could be changed to the AA state, if bandwidth estimation represents the base layer, which also could not be supported. A timer is used in the AA state; if this timer expires, the connection is closed.

The approaches [160, 161] uses a play buffer, and provide more flexibility to the application, but may suffer from unexpected stops in playing, caused by lack of buffered data. The major reason for this buffer underflow is a mismatch between receiving rate and decoding data rate due to irregular network conditions. In order to resolve this issue, the following approaches provide ways to estimate the initial buffer, which is enough to avoid buffer underflow. However, this introduces delay.

**Xu and Helzer** [160] proposed the algorithm which estimates average playback rate and rebuffering probability of given TFRC connections with Markov-Renewal Modulated Deterministic Process (MRMDP) for TFRC traffic. With a CBR source, m-MRMDP provides $m$ states presenting different TFRC rates. Since $m$ is an even number, media rate $S$ divides the states into two groups, which have an equal number of higher and lower rate ranges. In order to validate the model, initial buffer and rebuffering events are simulated. It is assumed that video encoding is CBR and a single-source streaming approach is adopted.

**Predictive Buffering Algorithm (PBA)** [161] for Multiple-source Streaming assumes that the bandwidth of each connection from sender to receiver is an independent random variable. Thanks to the Central Limit Theorem, the aggregated bandwidth can be a normal distribution. With these characteristics, the problem could be simplified as getting precise average and variance of aggregated bandwidth. In order to measure these precisely, the authors use a confidential interval, which gives more reliability to an estimation of initial buffer.

### 3.5.5 Summary

This section discussed related real-time multimedia delivery schemes. Firstly, a services type based categorisation was discussed including broadcasting and on-demand. There are various approaches using multicast or unicast for both service types. Smart PIN focuses on the on-demand approach which utilises multiple-source streaming. To account for quality of delivery, QoS support during streaming is also discussed. Some classic adaptation schemes which control multimedia rates are presented. In addition, buffer prediction approaches are also listed.

## 3.6 User Utility Function

Utility functions are mainly used in Economics for presenting the relative satisfaction from consumption of goods or services [163]. After their success, they were also used in different other fields including networking in order to reflect user satisfaction with the service based on measurable metrics from the system. In this section, a high level description of the utility and related approaches in various areas of network is introduced.

Usually, a utility function could be written as in Eq. 3.1 with a 0 to 1 range, where $X_1, \ldots X_n$ represent variables. The variables used in a utility function indicate resources, parameters, preferences, content, etc. The shape of commonly used utility functions could be exponential, linear, min-max, linear piece-wise or discrete according to their application [164, 165]. More simply, a linear utility function could be presented as Eq. 3.2, whereas $i$ is from 1 to $n$ and $w_i$ is the weight of each parameter $X_i$. Also, $\sum_{i=1}^{n} w_i = 1$.

$$U = f(X_1, X_2, \ldots X_n) \tag{3.1}$$

$$U = w_1 \cdot X_1 + w_2 \cdot X_2 + \ldots + w_n \cdot X_n = \sum_{i=1}^{n} (w_i \cdot X_i) \qquad (3.2)$$

For general network applications, a utility function is used in network pricing schemes [166, 165], network selection (or handover) [167, 168], etc. MacKie-Mason and Varian propose a pricing scheme for congested network resources such as FTP or Web servers, using an economic model. Based on the competitive market concept, they show equilibrium of price and capacity which maximise net social benefits. Wang and Schulzrinne [165] proposed two pricing schemes working with utility function based on user preferences. Ylitalo et al. [167] propose a solution for facilitating a user making a network interface selection decision, whereas Zhu and McNair [168] propose a vertical handover scheme based on a cost function consisting of QoS.

For data applications, some approaches present utility functions and use them for designing a game policy. The cost function used by Chun et al. [61] consists of storage cost and production of delivery cost and user demand. Geomans et al. [129] use a benefit function presented with user demands and number of serving node. If these two cases are consider together, utility functions relating content could be determined with cost and benefit for as shown in Eq. 3.3, whereas $U_i$ is utility of content $i$, $B_j$ is benefit of parameter $j$, and $C_k$ is cost of parameter $k$. One interesting is most of approach consider user demand as popularity rather than separate user's interest.

$$U_i = (\sum w_j \cdot B_j) + (\sum w_k \cdot C_k) \qquad (3.3)$$

The utility function reflecting user relative satisfaction could be presented with benefit and cost, even though it considers only a few factors related to the application. Since Smart PIN targets a performance and cost-oriented user utility based approach, a utility function is used for assessing the value of the operation which requires an autonomous decision.

## 3.7 Chapter Summary

Next Generation Networks (NGNs), person-centric context-aware systems and data replication systems were introduced in this chapter. NGN approaches assume a heterogeneous network environment. However, most of the solutions focus on network issues and there is a lack of consideration of performance and cost oriented approaches in terms of users, their device and their contents. Unlike person-centric context-aware systems which only consider exchange of small size context data, the Smart PIN uses large size files and focuses on performance and cost of content delivery. To support large amounts of data and to consider user utility, Smart PIN needs to use metadata for data, performance and cost metrics for network and system. User utility could be represented with a benefit and cost. Smart PIN considers not only non-realtime applications but also realtime applications for multimedia supporting a cooperative manner.

# Chapter 4

# Smart Personal Information Network

## 4.1 Introduction

Smart PIN is a novel performance and cost-oriented context-aware personal information network which enables efficient user access to information located on remote devices. Smart PIN operates on heterogeneous network environments and considers user, network and device characteristics. It also takes into consideration delivery cost during communication.

In Smart PIN it is assumed that the same categories of informational context and content exist in distributed sources. For example, users could have emails which are stored on a desktop at their house, on a laptop at work and on webmail as part of a web portal service. For user convenience, the integration of these services is very desirable. Consequently, Smart PIN adopts data replication as one of its basic principle.

In order to cope with the information overload, context-awareness should be considered. However, most of the systems such as MMM2 [32] and OmniStore [63, 64] just consider context annotation and do not consider automatic data duplication based on performance efficiency. Furthermore, data duplication systems such as OceanStore [65], Wayfinder [70] and OmniStore [63, 64] just focus on homogeneous network interfaces within a specific topology such as wireless or fixed network. The application level adaptation model used by

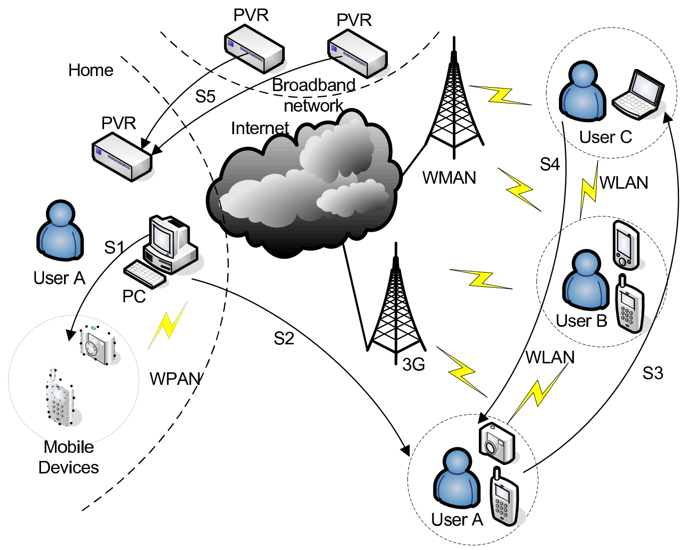



Figure 4.1: Overall concept of Smart PIN

Smart PIN supports performance, cost-effectiveness and a certain level of quality of service.

Due to continuous changing network topology, Smart PIN supports robustness of service management. Smart PIN should work not only with devices capable of heterogeneous network connectivity, but also with those devices supporting a single wireless network technology. Similar to the usual PAN-based solutions, it is envisaged that multiple Smart PINs could co-operate with each other for exchanging not only context and content, but also for using sensors and services.

As discussed in section 1.1, Fig. 4.1 illustrated Smart PIN concepts. It can be seen how different users A, B, and C located in a heterogeneous network environment have access to a number devices, which store content. Smart PIN enables content exchanges between these users based on their interest in it, while considering network and devices characteristics and aims at increasing delivery performance.

Smart PIN works in user-centric networks such as WPAN and WLAN, and concentrates on application-level abstraction of each layer in the system. This approach enables utilisation of user-centric parameters such as user utility in the applications of Smart PIN. The following sections will discuss the **design of Smart PIN**, a newly proposed **data replication scheme** which supports efficient user access to information located on remotely distributed devices in a heterogeneous network environment, and a novel **multiple-source multimedia delivery scheme** to achieve high multimedia quality without content adaptation to network conditions.

## 4.2 Smart PIN System Design

### 4.2.1 System Architecture

As Fig. 4.2 schematically presents, Smart PIN is composed of a number of interconnected distributed devices. At each device, three major components are deployed: Network, Service and Management.

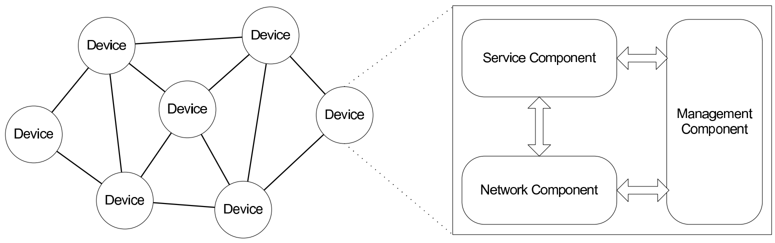



Figure 4.2: Smart PIN System Architecture

**The network component** covers vertical homogeneous interactions between the physical and link network layers for each technology. It is also concerned with heterogeneous network interaction through network layer bridges. Furthermore, it looks at automatic configuration and self-management in heterogeneous networks. Smart PIN involves a virtual agent in order to invite external services (e.g. Blogging) into the system. Support for personal privacy and security can also be provided. Auto-configuration and Self-management of Personal Area Networks (ASPAN) [169] and IEEE 802.21[1] Media Independent Handover (MIH) are possible approaches for automation of network layers that can be employed. However this needs to be considered in relation to the other components.

**The service component** focuses on service discovery, service composition, search and delivery for context and content, and access control protocols. The services need to be

[1] IEEE 802.21, http://www.ieee802.org/21/, last accessed 18 Nov. 2009

integrated and this service integration is responsible for searching distributed sources for digital items, managing integration of services and responding to client requests for integrated services. Applications search and transport the context and contents through the usual file transport protocols for integrated services. For overall services, various access controls could be applied for maintaining security and user privacy.

The goal of the **management component** is to support cost effectiveness in various aspects of the system that operate in a heterogeneous network environment. Cross-layer approaches such as [170] could be considered here. The context and status of networks and services are gathered in a management component and other components can use that for QoS, power efficiency or service differentiation.

### 4.2.2 Service Architecture

As an context-aware system, Smart PIN focuses on removing any unnecessary user interaction, while providing performance and cost benefits. Smart PIN assumes that the context will be presented via metadata using mark-up languages such as RDF[2], OWL[3], MPEG-7[4], MPEG-21[5] or combinations of these [171]. These standards describe the content stored on distributed devices and can be used for various applications. In particular, Smart PIN handles data pairs that consist of informational **context (or metadata) and content**, as described by the digital items in MPEG-21 [51].

Fig 4.3 presents different categories of services in Smart PIN. As we discussed in section 2.3, there could be **physical sensors** providing context from the actual devices such as location, environment, movement, etc., **virtual sensors** for context from applications, and **logical sensors** which generate enhanced context using different contexts from the previous two sensors. These sensors interact with Smart PIN services categorised into: primary,

[2]Resource Description Framework (RDF) / W3C Semantic Web Activity, http://www.w3.org/RDF/, last accessed 18 Nov. 2009

[3]OWL Web Ontology Language Overview, http://www.w3.org/TR/owl-features/, last accessed 18 Nov. 2009

[4]MPEG-7 Overview, http://www.chiariglione.org/mpeg/standards/mpeg-7/mpeg-7.htm, last accessed 18 Nov. 2009

[5]MPEG-21, http://www.chiariglione.org/mpeg/standards/mpeg-21/mpeg-21.htm, last accessed 18 Nov. 2009

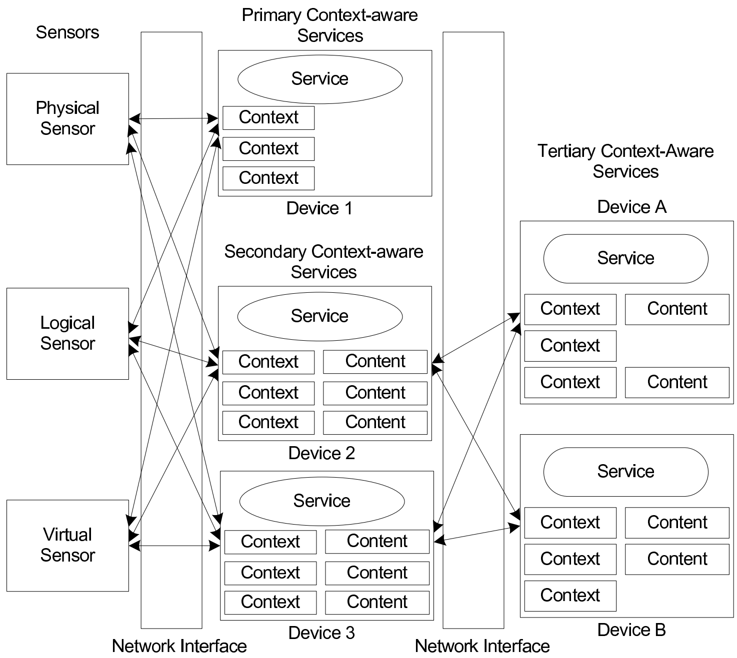



Figure 4.3: Smart PIN Service Architecture

secondary and tertiary services. A detailed discussion on each service follows.

- Primary Services: These services gather related context and provide it to the users directly. Usually, there is no extra generation of data. Sightseeing Assistant [31] is an example of such a service where the location context is grasped and used directly in the application.

- Secondary Services: The services generate user data with related context data. Usually, they annotate context on data for specific purposes. Mobile Multimedia Metadata 2 [32], OmniStore [63], hierarchical annotation of photo collections [172] are good examples of this. These systems gather the context and annotate it to the related content such as pictures and files.

- Tertiary Services: These services merge distributed secondary services and exploit distributed file systems or file replication services. Digitised pictures and videos (content) and context exist in the creator devices (e.g. digital camera) and storage devices (e.g. PC or personal digital recorder) and they can be accessed by users from mobile devices (e.g. a portable multimedia player). Wayfinder [70] and EnsemBlue [71] provide such services. Even though these solution do not directly use context, they use metadata for efficient processing or sharing of information.

Usually, primary services are most straightforward and popular approaches which use context information. They just collect the context and process those for the purposes of applications. Other two services are currently emerging as context-awareness systems. Especially, tertiary service has big potential since it supports a reasonable solution for resolving information overload.

Currently, Smart PIN is focusing on tertiary services. There are two extreme models for tertiary services: centralised and distributed. In the case of the **centralised model**, context is stored in a central agent even if content is distributed across agents and servers. Users can expect fast response when browsing with the centralised model. Furthermore, this approach can provide easy maintenance, but the server might suffer from high processing overhead. In the **distributed model**, context is distributed across the agents. This case is more robust and resilient in comparison with the centralised model, and has a relatively light synchronisation burden for the metadata. As discussed in section 3.4.2.1, a hybrid model is also possible for taking advantage of the benefits of both centralised and distributed approaches.

### 4.2.3 Operational Scenarios

The scenarios examined here are related to the service component assuming that network support is provided. Data used in the scenarios is represented in form of data pairs which include context and content. In this situation, access to context and content (the data pair) is possible after services are found and integrated; subsequently available data is composed. There can be several scenarios as described in section 1.1: data duplication to mobile devices (Situation 1), remote device access using mobile devices (Situation 2), data sharing with a user's colleagues (Situation 3), data sharing from a user's colleagues (Situation 4) and streaming from multiple devices (Situation 5). Although each situation looks different, each contains hidden steps which could be same in different scenarios.

- **Service status change**: This step is for handling movement of devices, or change of services. For example, a device joins an integrated service assuming it is not already involved in an integrated service or is integrated within another service. Another scenario involves a device leaving an integrated service. Furthermore, several devices could be connected via different networks. If there is extra available connectivity between these networks, these devices could join an integrated service. These issues will be discussed in detail in the service discovery and service composition management section later on.

- **Content access procedures**: This step defines how to search the content that a user wants and whether the user can use that content or not. This is actually based on an integrated service, which could be considered as an overlay network (e.g. P2P network). Smart PIN also considers sharing of content among user's devices and among various users.

- **Replication management**: This step determines which data will be replicated and where the data will be stored. In an integrated service, new data can be produced via direct generation or transfer from other devices. The deletion of existing data is also possible. A deletion scheme such as [63] takes care of deleting the least demanding duplicated data when it is necessary. Not only must we consider creation and deletion of data in one device, but we must also consider available data expansion, induced from integrating services and this is another significant scenario resulting from a service status change. These issues will be discussed in detail in the data replication strategy.

- **Data pair delivery**: This step is related to transfer of context and content data pairs. Smart PIN measures various parameters to determine when data pairs will be transferred to a specified node. This issue will be discussed in detail when the novel replication scheme for non-realtime data delivery and the new multiple-source streaming solution for realtime data delivery are introduced.

- **Content presentation**: This step is for presentation of delivered content to the user and processing of feedback from the user. Because of the properties of different devices, user preferences, etc, content should support adaptive presentation. During the presentation of content, the user may also want to control it with feedback interactively.

The main contributions of Smart PIN are related to **replication management** and **Data pair delivery**. In addition, some parts of **Content presentation** are also involved for multimedia delivery. The content presentation and other steps are discussed in the following sections, including the assumptions Smart PIN makes.

#### 4.2.3.1 Service Discovery and Service Composition Management

For handling service state changes, the services in mobile devices need to be found first. There are several proposed protocols like Service Discovery Protocol (SDP) for Bluetooth[6] and Simple Service Discovery Protocol (SSDP) for UPnP [173] which focus on **service discovery**. Existing service discovery mechanisms usually adopt a broadcast message such as WiFi beacon, advertisement and solicitation IP messages, etc. On the other hand, the network component in charge of the remote device requires a unicast based service discovery mechanism since the connectivity used between those is not appropriate for broadcasting. In this context, the selection of a proper protocol type needs to be considered next, assessing the characteristics of the network components layer.

When services are discovered, the device should be involved in **service composition** to provide an integrated service. Furthermore, a device can leave the network at any time and lose the connection, and the remaining devices should update their status. Briefly, service composition is concerned with building up an integrated service from basic services found through service discovery. In this context, this issue has quite a deep relationship with net-

[6]Bluetooth SIG, http://www.bluetooth.com/bluetooth/, last accessed 18 Nov. 2009

work composition caused by device inclusion/exclusion and network merge/separation scenarios. A good example for network composition is introduced in Service Specific Overlay Network (SSON) [174]. Auto-configuration and Self-management of Personal Area Networks (ASPAN) [169] also presents dynamic and automated network layer management on an Ambient Network. These schemes assumes that all devices have the same services which enables management of network and service composition at the same time.

Smart PIN supposes that service discovery is performed in the service component which is located on top of the IP layer of the network component. However, the layers below the network layer can contribute to this feature through a management component using a cross-layer approach. In addition, structured or unstructured P2P overlay management is used. However, service composition based on a P2P network is assumed in this thesis. Device availability can be measured using distributed manner [123] or a sever. It is assumed that there is a server which measures device availability for data replication.

#### 4.2.3.2 Content Access Procedures

Based on a composed network, there are required features in order for the user to access the content, such as search and access control. A searching algorithm provides the user with the facility for finding content which is wanted. On the other hand, access control defines whether that content is accessible according to the setting which is defined by its owner.

In the central server and client model, data access is relatively straightforward as the server has all information on the content. Clients which want to search for content need to send a query to the server, and then they can get response on the content. Some P2P approaches such as BitTorrent[7] use an approach similar to this. Usually, distributed approaches based on P2P include various kinds of algorithms for search. First of all, these could be categorised according to the P2P architecture such as unstructured or structured. In the case of unstructured P2P overlay, there is no rule for storing key information on content. Because of this, search algorithms use exhaustive ways such as flooding [175], depth-first

[7] The BitTorrent Protocol Specification, http://www.bittorrent.org/beps/bep_0003.html, last accessed 18 Nov. 2009

search [176] or breadth-first search [5]. However, structured P2P overlay supports indexing mechanisms such as DHT (e.g. CAN [177], Pastry [147], Chord [178], etc.).

In terms of accessing content, access control means who has the right to use content. The most popular way for access control is that all content has its own access control list which describes access authority. This is usually applied to the file system for operating systems such as NTFS for Microsoft Windows. However, access control should follow a model such as the discretionary access control [179] in distributed environments. Using distributed access control approaches, shared contents could be used according to the access rights as the owners decide for their data.

As mentioned earlier, Smart PIN does not specify which P2P structure is used, but it assumes that search features are supported. Another assumption in Smart PIN is that there are private content and shared content. Private content are shared among the user's devices. On the other hand, shared content will be disseminated to other user's devices for efficiency using distributed access control.

#### 4.2.3.3 Content Presentation

As devices vary significantly in characteristics, different multimedia presentation environments can be envisaged at different times. A user can see a movie clip on a mobile phone when he leaves from the office and then he might want to see the clip again on a HDTV when he arrives home. For this, adaptive multimedia presentation should be supported in Smart PIN, performing background and foreground procedures as depicted in Fig. 4.4.

The **foreground procedure** refers to the multimedia presentation and involves streaming the data among the nodes, adapting the user model and different device profiles. Interactive feedback such as control commands for multimedia adaptation (i.e. fast forward and rewind) and pre-defined feedback such as device profile and parameters can be used to fine tune the streaming to the network, so that the user receives a good multimedia presentation. Furthermore, content adaptation such as the digital item adaptation [51], help the data be reduced to an affordable amount for the mobile environment since these enable the control

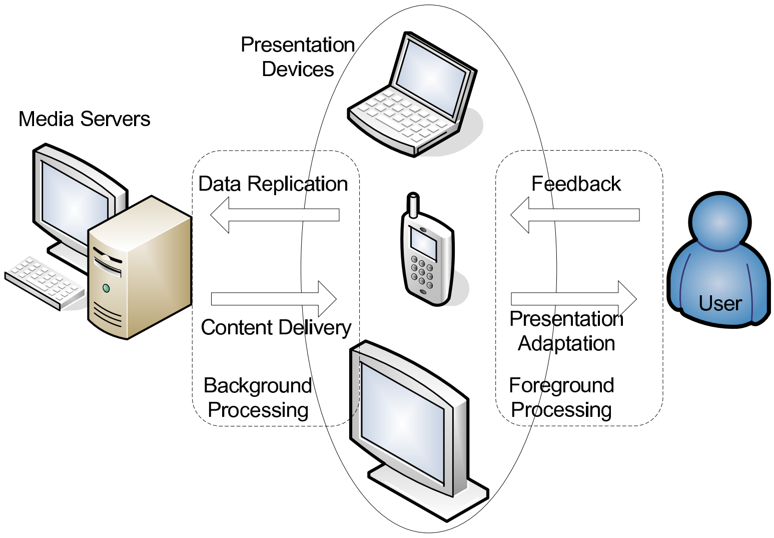



Figure 4.4: Adaptive multimedia presentation

of run time and quality factors, maximising user satisfaction.

The **background procedure** for the adaptive multimedia presentation performs data replication and data delivery. In order to handle large size non-real time data items, fixed-length segmentation (FIX_SEG) is used in a similar fashion to OceanStore [65], Wayfinder [70] and BitTorrent[8]. For multimedia data, variable-length segmentation (VAR_SEG) is employed to enable distributed streaming similar to VMesh [155].

As discussed, Smart PIN assumes that basic interaction for realtime multimedia is provided as user feedback. The presentation adaptation is involved with multiple-source streaming, as it cooperates with data delivery for realtime multimedia data.

### 4.2.4 Summary

This section presents the Smart PIN system and service architectures. The system architecture introduces network, service and management components. The service architecture shows that Smart PIN focuses on content sharing as a context-aware system in this thesis. For more detail, the operational scenarios introduced in chapter 1 was revisited, and the

[8]The BitTorrent Protocol Specification, http://www.bittorrent.org/beps/bep_0003.html, last accessed 18 Nov. 2009

steps used by the service component were described.

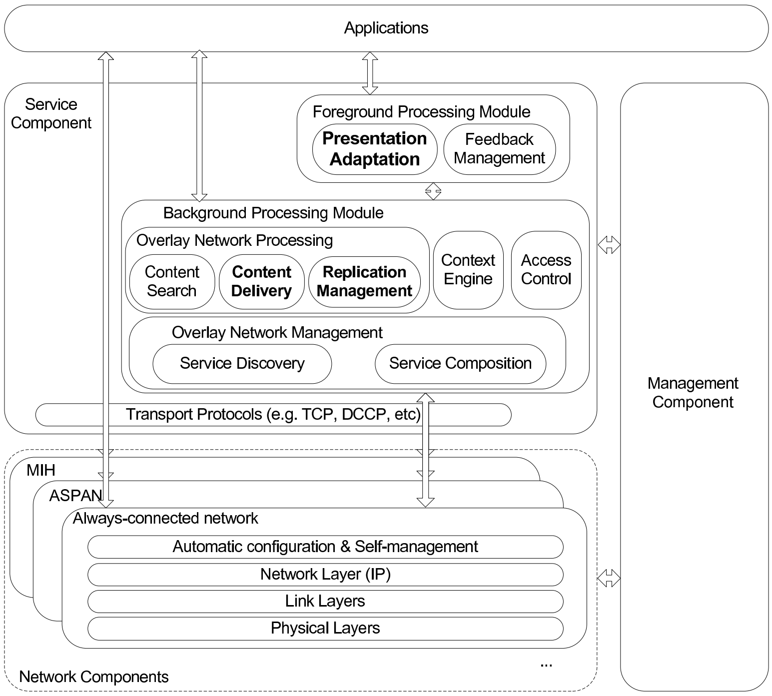



Figure 4.5: Detail Architecture of Smart PIN

The discussed design is summarised in Fig. 4.5. The network components suppose existing approaches such as MIH, ASPAN or other always-connected networks are used. The service component includes several assumptions including that overlay network management is provided, a content search function is supported, context annotated content is used, access control supports private content and shared content based on the user defined permissions. In addition, basic feedback management is provided.

The service component includes the main contributions of Smart PIN, which are data replication decision, content delivery and content presentation. Specifically, **novel data replication** and **quality adaptive multiple-source delivery schemes** will be introduced in the next sections.

## 4.3 Multimedia Data Replication Scheme

### 4.3.1 Introduction

Smart PIN is a performance and cost-oriented context-aware personal information network which focuses on efficient user access to information located on remotely distributed devices in a heterogeneous network environment. The dynamic characteristics of wireless networks strongly affect distributed application systems as nodes storing sharing data can get out of range and suddenly become unavailable. The nodes can become inaccessible also due to other factors related to their functionality: can be switched off, run out of battery, stopped working, etc. However, it is required to enable data replication in order to cope with these issues and enable full access to data.

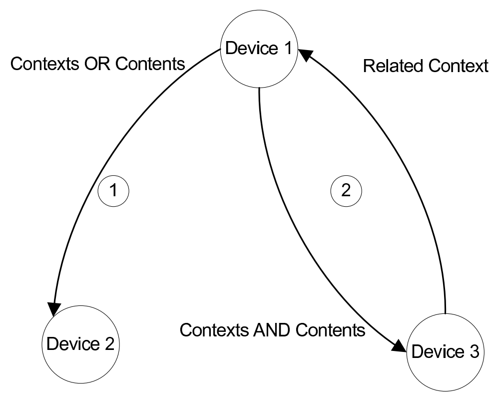



Figure 4.6: Types of data delivery in Smart PIN

As mentioned, Smart PIN considers data grouped in context-content pairs. Fig. 4.6 shows the main steps in distributing these data pairs. Usually, a device pulls data identified by a specific context (② in Fig. 4.6) which could be specified with several keywords. If the user has a certain level of interest in a piece of content, both context and content are transferred, otherwise only the context is delivered. Alternatively, context or context-content data could be pushed to specific devices (① in Fig. 4.6). If there is a device which has large storage space, a data replication scheme could be considered as a push-based approach.

In order to handle large-sized multimedia content, Smart PIN employs data segmentation into fixed length segments (FIX_SEG) and variable length segments (VAR_SEG). Small sized data is not segmented and is labelled NO_SEG. VAR_SEG data usually is data which uses realtime delivery such as a movie, and FIX_SEG is for other types of content.

Data replication is usually the main solution in order to achieve high data availability. In order to address both information overload, and the heterogeneity of devices and network connectivity, Smart PIN supports a utility function-based data replication scheme. In this context, Smart PIN employs a novel **Multimedia Data Replication Scheme (MDRS)**, which is proposed here and is described next. MDRS is divided into two steps: data selection and data delivery, and uses an utility function.

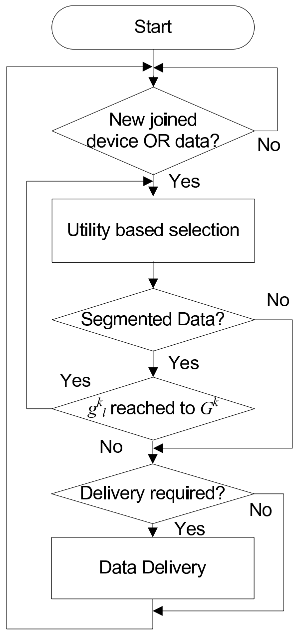



Figure 4.7: Data replication flow chart

Fig. 4.7 presents the MDRS algorithm. During data selection, data is classified into three categories based on two thresholds depending on their utility to the users. This categorisation determines which data will be replicated to other devices. In order to decide how many data sets are needed among the devices, a minimum data set requirement ($G^k$) is applied.

Smart PIN uses a proactive approach based on pull data delivery using the introduced utility function, in order to control network usage. When the system selects data to be replicated, it also decides on data delivery based on the utility function. Smart PIN calculates the transfer duration with the target bandwidth consumption and schedules data replication accordingly.

For a more detailed description, the following sections will discuss the utility function, data selection, minimum data set requirements and data pair delivery. As a deployment of the data replication scheme, an approach using pre-recorded contents of IPTV is introduced as well. Finally, the summary of this section follows.

### 4.3.2 Utility function

Utility functions are mainly used in the field of Economics for presenting the relative satisfaction from consumption of goods. However, recently they have also been used in other areas including networking to reflect users' satisfaction with services based on measurable metrics from the system. For data application systems, there have been some approaches which present and use utility functions, such as [61, 129].

Smart PIN bases its functionality on a utility function which includes two main components: the **private utility component** which reflects the user's individual interest in the content $i$ and the **global utility component** which expresses the overall utility of the content in relation to its popularity.

The private utility formula includes the content $i$'s associated **benefit ($B_i$)** and **its cost ($C_i$)** to the user. In addition, the **user interest** in the particular data item $i$ ($I_i$) (e.g. relevance of data to the user) is used to increase or decrease the relative influence of the benefit in comparison with that of the cost. Including normalised values of these metrics, the private utility ($PU_i$) for item $i$ is computed as in Eq. 4.1 and because it is normalised, it has values from 0 to 1.

$$PU_i = \frac{1 + B_i \cdot I_i - C_i}{2} \tag{4.1}$$

The global utility component formula ($GU_i$) includes — apart from the content $i$'s associated **benefit** ($B_i$) and **its cost** ($C_i$) — **the popularity** of the multimedia streaming segment ($G_i$) [155] as described in Eq. 4.2.

$$GU_i = \frac{1 + B_i \cdot G_i - C_i}{2} \tag{4.2}$$

The overall utility function which includes both $PU_i$ and $GU_i$ is presented in Eq. 4.3. Different weights are used depending on the data type ($T_i$). As Smart PIN assumes that there is both private and shared data, the weight of the overall utility function is also related to this. One example of possible weights is introduced in Table 4.1. Private data which is not shared is controlled by the private utility. The global utility fully controls the content which is not owned but which is shared. In case of owned and shared data, the total utility function is involved.

$$U_i = w_{1,T_i} \cdot PU_i + w_{2,T_i} \cdot GU_i \tag{4.3}$$

| Data types($T_i$) | Owned private | Owned shared | Not owned but shared |
|---|---|---|---|
| $w_1$ | 1 | 0.5 | 0 |
| $w_2$ | 0 | 0.5 | 1 |

Table 4.1: Examples of weights for the utility function

#### 4.3.2.1 Benefit and Cost Function Design

Smart PIN uses separate utility functions for data selection and delivery respectively and consequently the benefit factor used ($B_i$) will differ. In addition, various parameters which are logical and physical parameters could be classified and used for both benefit and cost functions, for example, access cost (e.g. free, fixed rate or packet based charges), energy consumption, bandwidth, memory usage, etc. The following example just includes memory and bandwidth for simplification, but extension are possible.

For data selection, the relevance score of content item $i$ ($B_i$) is used as **the benefit factor** ($B_{RL_i}$) representing the perceived quality of information in terms of the user's information, which is related to memory usage. In other words, users have $B_{RL_i}$ benefit if the content item $i$ is located in their devices. In contrast, for data delivery, the required bandwidth for the delivery of data item $i$ is used ($B_{BW_i}$) divided by the total available bandwidth for content delivery ($B_{BW}$). $B_{BW_i}$ is represented as $S_i/T_{dur_i}$, where $S_i$ is the size of item $i$ and $T_{dur_i}$ is the required duration of item $i$ delivery. Consequently, the benefit function is defined as in Eq. 4.4.

$$B_i = \begin{cases} B_{RL_i} & \text{For data selection} \\ \frac{B_{BW_i}}{B_{BW}} & \text{For data delivery} \end{cases} \tag{4.4}$$

A utility function's **cost component** considers the storage and delivery cost relative to the size of data as shown in Eq. 4.5. To normalise this factor, the maximum data size ($S_{MAX}$) and the minimum data size ($S_{MIN}$) are measured in a node which performs data replication on another node. The data selection and delivery scheme adopts the following cost factor for the utility function, where $S_i$ is the size of item $i$.

$$C_i = \frac{S_i - S_{MIN}}{S_{MAX} - S_{MIN}} \tag{4.5}$$

#### 4.3.2.2 User Interest Modelling

Since the mobile device can not bring all data available to the user, data most important to the user needs, should be stored on the mobile device itself. Although heterogeneous networks provide always-connected services, the user needs to pay certain costs such as data delivery time, communication tariff, additional usage of battery power, etc. This supports the need for data replication to adopt a utility function which includes benefit, cost and user interests as presented in Eq. 4.3. Benefit and cost could be measured from the system (e.g. storage cost, delivery cost, etc.) However, the eagerness of the user as a factor of utility function needs to reflect the actual user interest. Using a user model and associated metadata, this is feasible supporting user-awareness.

A user modelling system collects different types of information (e.g. user's interests)

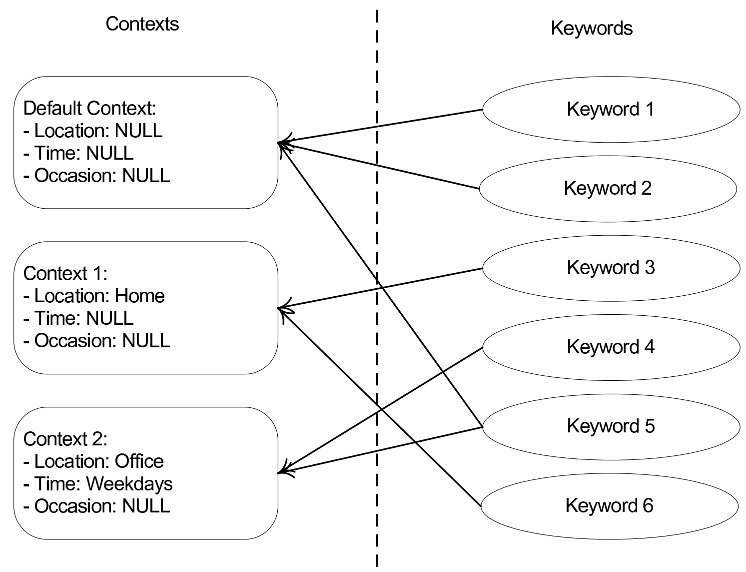



Figure 4.8: User model presented with context and keyword pairs

about the current user and performs adaptation and other operations based on the analysis of this information [180]. In Smart PIN, this could be simply collected through the user's previous queries in a specific context or environment such as location, time, etc. This user model could be represented as the context — keyword pairs as illustrated in Fig 4.8. Another simple user model could be a actual user profile [181], which stores a set of keywords given by the user. No matter what particular user model Smart PIN adopts, data replication needs to use it in order to handle and represent the user interest. In this work, user modelling is not a focal issue for Smart PIN but it is an important part of the basic concept of the overall system.

With given keywords from a user, devices in Smart PIN can evaluate how important each data item is to a user. One of the main issues which are discussed in Information Retrieval systems is related to document-query pair relevance assessment [182]. IR systems can provide relevance values against keywords and data. Data with high relevance scores with query keywords from the user model will be considered as data likely to be more important to the user. Simple metadata and keyword comparisons also provides a primitive way to measure relevance of data.

#### 4.3.2.3 Global Popularity

The popularity of a segment of distributed content is assumed that has a Zipf-like distribution [143, 155]. If all the segments are ranked in descending order of their popularity, and the popularity of the $i$th segment, $p_i$, is expressed as

$$p_i = \frac{1/i^\alpha}{\sum_{n=1}^{N} 1/n^\alpha} \tag{4.6}$$

where $N$ is the total number of segments, and $\alpha$ is a constant which is characterising the distribution.

In order to measure the actual popularity of a segment, statistics on user requests are used. If there is a central server, user requests could be easily collected for this purpose but it is not easy to do this using a distributed approach. However, VMesh provides a distributed popularity estimation scheme for segments of distributed contents [155]. Based on the assumption that this can be used, Smart PIN uses the popularity of segments of content for the global utility function.

### 4.3.3 Data Selection

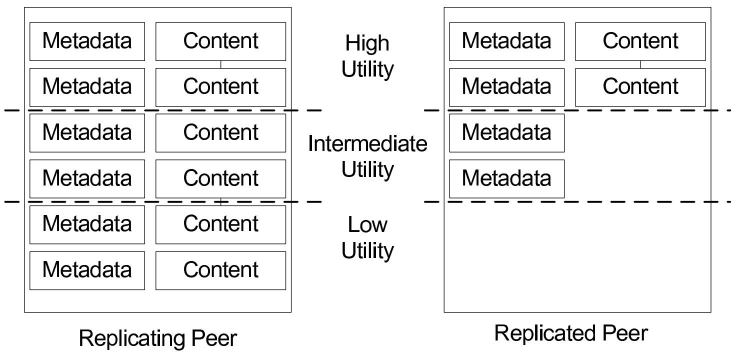



Figure 4.9: Data replication using classification

Based on utility function values, during data selection, data is classified into one of three categories based on two thresholds as presented in Fig 4.9: high utility ($U_{HU}$), intermediate utility ($U_{IU}$), and low utility ($U_{LU}$). Data with high utility ($U_i \geq U_{HU}$) will be replicated onto devices as a metadata-content pair. As a user might want to access information with intermediate utility levels ($U_{HU} > U_i \geq U_{IU}$), in this case only the metadata will be replicated onto the device, offering information about the location of data if it needs to be transferred. Data with low utility values ($U_i < U_{IU}$) will not trigger an operation, saving network and storage resources. More detailed categorisation including delivering a part of the content, a part of context, etc. can be considered, but Smart PIN considers currently only these three.

If data is selected, a node is chosen to store the data for replication. In order to improve data availability, Smart PIN selects the nodes which have high device availability, since data availability is highly dependent on device availability [123]. Among devices which have the same device availability, Smart PIN selects a node randomly as it is the easiest and most efficient way.

### 4.3.4 Minimum Data Set Requirement

Data replication is often used when data is available on distributed devices and their availability differs. The process refers to making copies of data in order to increase their availability to the overall system. In this context, it is not possible to guarantee a successful multimedia streaming process if there is not at least a full set of stable segments of multimedia data available. Therefore it is desirable to have data availability closest to 1.

The **availability of a device** ($P_j$) is defined as in Eq. 4.7. Eq. 4.8 defines the average data availability from the availability of the segment $l$ of the multimedia data item $k$ in a device $j$ ($s^k_{l,j}$), device availability ($P_j$), total number of segments of the multimedia data $k$ ($L$) and total number of devices ($J$).

$$P_j = \frac{Device\ j\ available\ time}{Total\ time} \tag{4.7}$$

$$D^k_{avg} = \frac{\sum_{l=1}^{L} \sum_{j=1}^{J} (s^k_{l,j} \cdot P_j)}{L \cdot J} \tag{4.8}$$

It can be shown mathematically that the system availability of the replicated segment $l$ of the multimedia data $k$ across all devices ($g_l^k$) is always greater than or equal to $s_{l,j}^k$ as Eq. 4.9 indicates. By combining Eq. 4.8 and 4.9, the relationship from Eq. 4.10 is derived.

$$s_{l,j}^k \leq \sum_{j=1}^{J} s_{l,j}^k = g_l^k \tag{4.9}$$

$$\begin{aligned} D_{avg}^k &\leq \frac{\sum_{l=1}^{L} \sum_{j=1}^{J} (g_l^k \cdot P_j)}{L \cdot J} \\ &= \left( \frac{\sum_{l=1}^{L} g_l^k}{L} \right) \cdot \left( \frac{\sum_{j=1}^{J} P_j}{J} \right) \end{aligned} \tag{4.10}$$

Denoting $P_{avg}^J = \frac{\sum_{j=1}^{J} P_j}{J}$ and $G_{avg}^{L,k} = \frac{\sum_{l=1}^{L} g_l^k}{L}$, Eq. 4.10 is simplified as Eq. 4.11.

$$D_{avg}^k \leq G_{avg}^{L,k} \cdot P_{avg}^J \tag{4.11}$$

$G_{avg}^{L,k}$ represents the average availability of multimedia data $k$ in Smart PIN and is dependent on the number of multimedia segment sets in the system. As data availability cannot exceed 1 and it is desired that the availability is as high as possible, Smart PIN aims to find the minimum number of sets of segments from the multimedia data $k$ such as for a given average device availability $P_{avg}$ to have the relationship as defined in Eq. 4.12. From equation, the $G^k$ target is derived as in Eq.4.13.

$$1 = G^k \cdot P_{avg} \tag{4.12}$$

$$G^k = \left\lceil \frac{1}{P_{avg}} \right\rceil \tag{4.13}$$

If $G^k$ exceeds the number of total devices ($J$), some nodes may include more than one duplicated instance of a segment for specific data. However, it is not desirable for a node to include same segments. Finally, $G^k$ can be defined as in Eq. 4.14.

$$G^k = min(J, \left\lceil \frac{1}{P_{avg}} \right\rceil) \tag{4.14}$$

Smart PIN utilises the minimum value for processing segmented multimedia data separately from the other data pieces in order to sustain the minimum set of multimedia segments among the devices. As mentioned, it is assumed that there is a server which measures device availability in the thesis. Based on the measured device availability ($P_{avg}$), the server can calculate $G^k$. Replication nodes can get $G^k$ from the server and use that for data replication. Smart PIN replicates each segment of the VAR_SEG data in order to have the average availability of multimedia data $k$ reach the target value of $G^k$ and therefore provide maximum data availability of data given a certain level of device availability. The transfer of the segments is based on the utility function.

### 4.3.5 Data Pair Delivery

Although the data selection decides which data will be replicated and where that data will be stored, the actual replication operation can occur during data pair delivery. Most schemes for data replication for mobile devices [63, 67, 65, 123] focus on the delivery and replication scheme for files but do not consider the cost of delivery. Although Tempo [126] decides data delivery to be based on a cost related parameter, the bandwidth budget, this approach does not consider user interest related to data. For the Smart PIN data, this section presents the delivery scheme which considers both user interest and costs.

As discussed earlier, data delivery in data replication could be reactive and proactive. In a reactive way, the system uses triggers for each user, for example, the system can set a trigger dependent on communication costs and budget. The communication budget is assumed that it increases as time goes by. If the budget is enough to cover the communication cost for a delivery, the system performs the data delivery. In a proactive way, the system can check periodically and decide on data delivery based on criteria which the user sets. In Smart PIN, the triggers for the reactive and proactive approaches are related to the utility function which includes cost as one of its factors.

Devices in Smart PIN have different characteristics in terms of device profile. The model needs to consider those characteristics for improving performance and efficiency. For example, if a powerful server is available, it is chosen as the master for the integrated service and consequently the mobile devices will benefit if most transmissions occur via the server. Alternatively, a dynamic approach utilising a centralised approach or hybrid peer-to-peer can be used.

The characteristics of data also influence data delivery costs. For example, large sized data might have higher costs than a smaller one. This information could be easily acquired from metadata which includes file attributes. Sometimes, the user might be willing to receive data even though the cost of delivery is quite high. Because of this, there should be consideration of user interest as a factor for data delivery.

Since different connectivity technologies have their own characteristics, the cost of communication varies in heterogeneous network environments. Simply, WMAN, WLAN and WPAN have different bandwidths [183]. Furthermore, power consumption is also different from each other because they use different technologies [184]. This cost might include not only physical costs but also logical costs related to tariff of the service, user preferences, and so on. As mentioned earlier, benefit component of the utility function for data delivery is defined with the required bandwidth for the delivery of data item $i$ is used ($B_{BW_i}$) and the total available bandwidth for content delivery ($B_{BW}$). Since $B_{BW_i}$ is represented as $S_i/T_{dur_i}$, the utility function derives $T_{dur_i}$ as described in Eq. 4.17 where ($S_i$) is the size of item $i$.

$$B_i = \frac{B_{BW_i}}{B_{BW}} = \frac{S_i}{B_{BW} \cdot T_{dur_i}} \tag{4.15}$$

When the private utility function only is involved, $T_{dur_i}$ can be retrieved as Eq. 4.17.

$$U_r = \frac{1 + \frac{S_i}{B_{BW} \cdot T_{dur_i}} \cdot I_i - C_i}{2} \tag{4.16}$$

$$T_{dur_i} = \frac{S_i \cdot I_i}{B_{BW}(2 \cdot U_i^r + C_i - 1)} \tag{4.17}$$

In this section, delivery schemes and cost of delivery are discussed. Among the characteristics of Smart PIN data delivery, the model of communication cost for data pair delivery will be discussed in following sections. For this, cost function design will be discussed. As an application based on our proposed cost function, a memory management scheme will be introduced.

### 4.3.6 Data Replication for IPTV

As discussed in section 2.2.3, IPTV backbone network includes the super head end (SHE) and a number of video hub offices (VHOs). VSOs are attached to access networks such as DSL. In order to apply our algorithm for data replication to the IPTV scenario, a system architecture for IPTV multimedia data sharing and replication is presented in Fig. 4.10. As shown, apart from VSO, VHO and SHE, a Cache Server (CS) is included. These cache and synchronise multimedia content for services such as VoD, recently broadcasted programs through Content Distribution Network (CDN), etc. [185]. Furthermore, VSO works with STBs in the User Group within the IP-based access network. Specifically, VSO and STBs use a P2P network and benefit from hashing features such as Distributed Hash Table (DHT). This thesis focuses on the operation of the User Group only and in particular when using variable length segmented (VAR_SEG) multimedia data and does not concern another aspects.

To maximise performance, the User Group keeps the minimum set of multimedia data segments, and data replication should be involved during recording of program $k$. A minimum number of data sets is defined in the previous section and is related to the number of recording nodes (e.g. STBs or Personal Video Recorders (PVRs)) for a program $k$ ($S_k(t)$). It is assumed that program $k$ starts broadcasting from time $t_{k,start}$ and ends at $t_{k,end}$. Before $t_{k,start}$, $S_k(t)$ represents the number of nodes which are actually recording the program $k$ between $t_{k,start}$ and $t_{k,end}$. After $t_{k,end}$, $S_k(t)$ indicates the number of nodes that share the

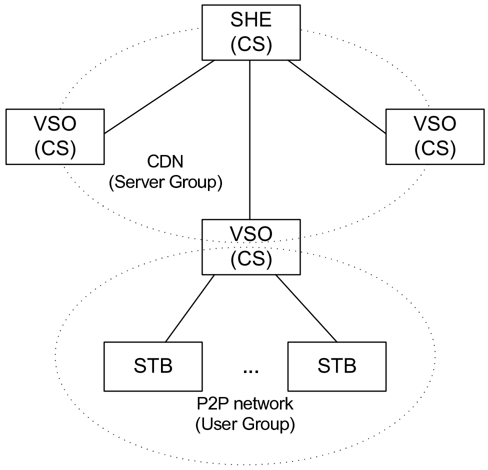



Figure 4.10: Proposed architecture for sharing and replication of IPTV multimedia data

recorded program $k$.Three possible cases for this are examined next.

#### 4.3.6.1 Case 1: $S_k(t) = 0$

CS at the level of VSO may cache all broadcast content if there are no recording nodes. However, there is a limitation of memory to store large amounts of data and because of this, CS just includes recent programs, and older recorded broadcast programs are discarded from storage.

Initially, $S_k(t)$ is 0 since there is no node to record the program $k$. If $S_k(t)$ remains 0, no recording is performed and no extra action is required.

#### 4.3.6.2 Case 2: $0 < S_k(t) \leq G^k$

When there are nodes to record a program $k$ (e.g. $S_k(t) > 0$), the minimum multimedia data segment set should be maintained as $G^k$. Until $G^k$ is met, replication should be performed. CS supports this replication if the program $k$ has already started (e.g. $t > t_{k,start}$). Replication is scheduled based on the Smart PIN algorithm which controls network load.

#### 4.3.6.3 Case 3: $S_k(t) > G^k$

This case indicates that the number of recording nodes exceed the minimum data sets required for maximum performance. Therefore no data replication is necessary at this stage. In order to save space, the surplus of data stored could be reduced, but is not the focus of this thesis.

### 4.3.7 Summary for Data Replication

Smart PIN adopts a novel **Multimedia Data Replication Scheme (MDRS)** algorithm to support efficient user access to information located on remotely distributed devices in a heterogeneous network environment. MDRS includes data selection and data pair delivery based on a utility function. It specifically supports minimum data set requirements which are based on device availability. During data selection, data is classified into three categories based on two thresholds depending on their utility to the users. This categorisation determines which data will be replicated to other devices. A proactive pull data delivery is involved for data replication and it supports control of network usage based on content utility. In order to envisage its application, pre-recorded content replication for IPTV system is also introduced using MDRS.

## 4.4 Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD)

### 4.4.1 Overview

The Smart Personal Information Network (Smart PIN) targets efficient user access to information located on remote devices based on a performance and cost-oriented approach. Smart PIN assumes its data will be stored in context-content data pairs and considers annotated context as part of metadata. Additionally, Smart PIN focuses on replication of context-annotated content which can be part of the non-realtime replication procedures as already described in section 4.3 or part of the realtime multimedia content distribution which will be discussed in this section. Figure 4.11 illustrate Smart PIN and the two mechanisms — data replications and multiple-source delivery. Smart PIN's multimedia delivery solution consists of background and foreground procedures. As mentioned earlier, data replication belongs to the background procedure. Multimedia delivery is part of the foreground procedure of Smart PIN.

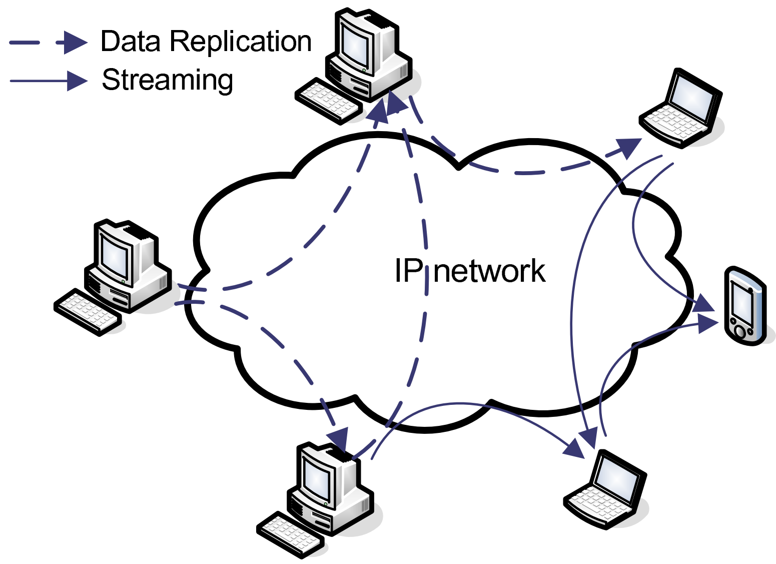



Figure 4.11: Smart PIN overview

There are two types of solutions for multimedia delivery in terms of the number sources: single and multiple source schemes. Compared to single source streaming [155], multiple sender based approaches show good performance when dealing with variable network conditions during actual multimedia streaming. Some of these multiple source approaches make use of overlay multicast [142, 143, 157]. A UDP based approach, the PROMISE architecture [143] presents a multi-path live streaming approach: CollectCast which enables the choice of best sender sets within the current conditions. However, for simplicity these schemes use a Constant Bit Rate (CBR) only when video streaming [158], unlike the real-life high quality video encoding and delivery which in general uses a Variable Bit Rate (VBR). In addition, they do not maintain high Quality of Service (QoS).

There are several approaches [156, 85] that use unicast based multiple source streaming. Among these, Nguyen and Zakhor propose Multiple Sender Distributed Video Streaming (MSDVS), a framework for streaming video simultaneously from multiple mirror sites to single receivers over the Internet [85]. In order to increase tolerance against loss and delay due to network congestion, they adopt a rate allocation and a packet partition algorithm. However, they also do not consider quality-related issues.

A significant problem in the unicast-based streaming is the mismatch between the available network bandwidth and media encoding/sending rates. There are two major avenues to solve this issue: media adaptation [86] and data buffering [160, 161]. Media adaptation approaches adjust multimedia quality to the available network resources. The approaches using data buffering provide more flexibility to applications, but often are affected in their streaming quality by buffer underflow. In general good buffer underflow management enables better user perceived quality.

This section describes the **Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD)**, a novel scheme which maintains high quality data delivery without media quality adaptation to network conditions. In order to overcome varying network conditions, QAMMD uses an innovative double buffering architecture which includes virtual multiple buffers associated with multiple network connections and the classic decoding buffer.

Fig. 4.12 present QAMMD algorithm. When it starts, there is single buffer estimation for each connections and multiple buffer estimation. If buffer includes enough data, QAMMD starts play of multimedia sequence. The next sections describe the QAMMD algorithm in detail. First of all, a double buffering architecture for QAMMD is introduced. For basic understanding, the single TFRC buffer estimation model is discussed. After this, the multiple buffer estimation model is presented based on the single TFRC buffer model. In addition, segmented data streaming is described based on distributed segments of content. Finally, the summary of this section follows.

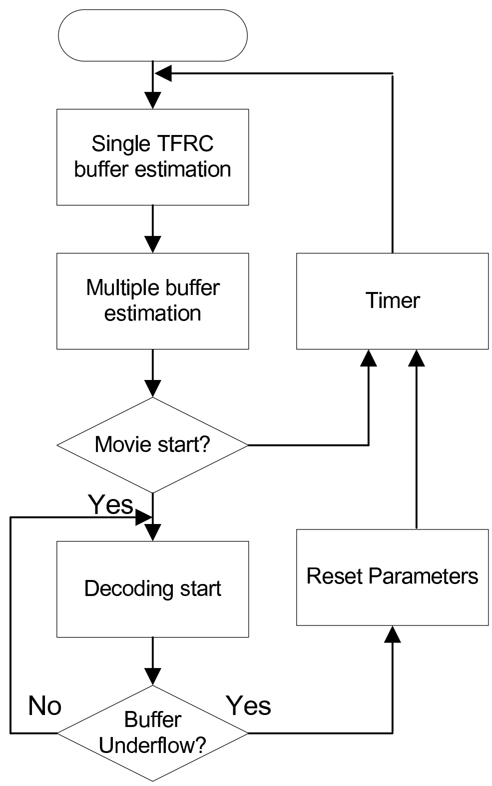



Figure 4.12: QAMMD flow chart

### 4.4.2 Double Buffering Architecture

In order to support the delivery of high quality multimedia streams, the proposed unicast-based multiple-source streaming approach, QAMMD adopts a novel double buffering architecture. It includes $n$ **senders**, $n$ **connections** from each streaming senders to a receiver and **two levels of buffers** at the receiver as shown in Fig. 4.13: **virtual multiple receiving buffers** and **a play buffer**. Additionally, a novel **Buffer Coordination Module (BCM)** balances the functionality of this two-level buffer structure. Currently, all the buffers on the receiver side are assumed unbounded in size and the senders are assumed to share same amount of multimedia data to be streamed to a receiver.

The virtual multiple receiving buffers are managed as **Individual Storage Spaces** (ISS). Each ISS stores multimedia data received via one of the $n$ connections established between the multiple sources and the receiver. Although other protocols can also be used for this

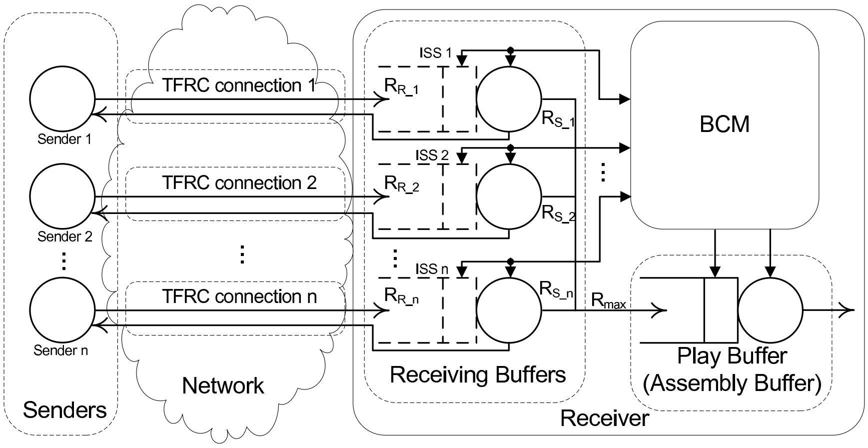



Figure 4.13: Example of a QAMMD-based Multimedia Delivery System

purpose, QAMMD makes use of the TCP Friendly Rate Control (TFRC) protocol [83], in order to best balance the aggressiveness of the multimedia delivery with friendliness towards other traffic. Each ISS $i$ receives data via the network with a rate, $R_{R\_i}$ and provides data to the play buffer with rate $R_{S\_i}$. $R_{R\_i}$ is estimated using TFRC throughput [83], whereas $R_{S\_i}$ is determined based on dividing the maximum media encoding rate ($R_{max}$) by the number of nodes. Although ISSs do not really store the data (the play buffer stores it for efficiency), they enable BCM control of the data flow to the play buffer for buffer underflow avoidance.

The **play buffer** uses the MPEG Video Buffering Verifier (VBV) mechanism [186] with an unbounded buffer size. When the number of packets in the play buffer reaches the initial number of packets set for efficient buffering ($S_{init}$), the data is fed to the decoder. This MPEG VBV operation guarantees that encoding-related factors do not cause buffer underflow in the play buffer given certain VBV buffer sizes, VBV delay and maximum media encoding rate $R_{max}$, as required by local playback [187]. In these conditions, the play buffer will receive data at the $R_{max}$ rate, which can be determined at encoding time. Consequently, $R_{max}$ is used as the aggregated target value for the overall ISS sending rate which is $\sum_{i=1}^{n} R_{S\_i}$. However, when setting $S_{init}$, current network conditions are considered, as remote delivery is often very different from local playback.

The **Buffer Coordination Module** (BCM) controls data flow between the TFRC connections, ISS and play buffer. BCM involves packet partition and rate allocation mechanisms which are discussed later. To manipulate buffer parameters based on the information from the buffers and multimedia data in order to balance the receiving buffers and play buffer, BCM retrieves media related information such as VBV buffer size, VBV delay, and media rate ($R_{max}$). In addition, it determines receiving buffer parameters such as $R_{S\_i}$, $S_{init}$, etc. In doing so, BCM uses an innovative **buffer underflow avoidance scheme (BUAS)** which is described in the next subsection.

### 4.4.3 Playing Buffer Underflow Estimation

Adopting a play buffer in the streaming application benefits quality of service as network condition varies. However, this approach may suffer from buffer underflow. As shown in Fig. 4.14, a single buffer receives data at the rate, $R_R$ and consumes data at the rate, $R_S$.

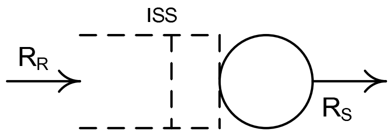



Figure 4.14: A single buffer from QAMMD-based Multimedia Delivery System

Xu and Helzer [160] model a single TFRC traffic which is similar to Markov Modulated Deterministic Process (MMDP), which is a popularly used ATM traffic model. They provide **buffer underflow probability (BUP)** functions as both a closed form and an iterative form, which means the total duration of all buffer underflow events is bigger than 0 sec. The closed-form BUP, $\gamma(x)$ is presented in Eq. 4.18, where $x$ is the number of initial buffering packets.

$$\gamma(x) \approx \left( F_{\hat{\theta}_n}(\hat{\theta}_{\text{media}}) + (1 - F_{\hat{\theta}_n}(\hat{\theta}_{\text{media}})) \frac{1 - e^{-\beta}}{1 - e^{-\alpha}} \right) \cdot e^{-(\alpha - \beta)x}, \quad x > 0 \tag{4.18}$$

As shown in Eq. 4.19, $F_{\hat{\theta}_n}(k)$ is the exponential random variable cumulative distribution function of $\hat{\theta}_n$ (i.e. the number of packets successfully sent between loss events) which is presented with loss event rate ($p$).

$$F_{\hat{\theta}_n}(k) \approx 1 - \sum_{h=0}^{6} \frac{(7pk)^h}{h!} \cdot e^{-7pk} \tag{4.19}$$

Eq. 4.20 is throughput function of TFRC, which is defined with loss event rate ($p$) and round trip time ($rtt$).

$$R(p, rtt) = \frac{1}{rtt \cdot \sqrt{\frac{2p}{3}} + 12 \cdot rtt \cdot \left(\sqrt{\frac{3p}{8}}\right) \cdot p \cdot (1 + 32p^2)} \tag{4.20}$$

As defined in [160], $\hat{\theta}_{\text{media}}$ is presented in Eq. 4.21, where $R_S$ is the consuming rate of delivered multimedia and $R(\cdot)$ is Eq. 4.20. Using Eq. 4.19 and Eq. 4.21, $F_{\hat{\theta}_n}$ can be presented as a function with $R_S$ and $rtt$.

$$\hat{\theta}_{\text{media}} = \frac{1}{R^{-1}(R_S, rtt)} \tag{4.21}$$

In addition, when achieved bitrates of single TFRC, $R_R$is provided, $p$ is also retrieved using Eq. 4.22.

$$p = R^{-1}(R_R, rtt) \tag{4.22}$$

$\alpha/\beta$ are the inverse of the expected value of changing buffered packets in decreasing/increasing states of play buffer, which are presented as Eq. 4.23 and Eq. 4.24, where $E[\hat{\theta}_{\text{media}}|S_n = i]$ is defined as Eq. 4.25, $q_{ij}$ is the state transition probability from the state $i$ to the state $j$, and $R_i$ is the receiving rate of state $i$.

$$\alpha = \frac{1}{\left(\left(\frac{1}{1-q_{00}} - 1\right) \cdot E[\hat{\theta}_{\text{media}}|S_n = 0] + 6(E[\hat{\theta}_{\text{S}}|S_n = 1] - E[\hat{\theta}_{\text{media}}|S_n = 0])\right) \cdot \left(\frac{R_S}{R_0} - 1\right)} \tag{4.23}$$

$$\beta = \frac{1}{\left(\left(\frac{1}{1-q_{11}} - 1\right) \cdot E[\hat{\theta}_{\text{media}}|S_n = 1]\right) \cdot \left(1 - \frac{R_S}{R_1}\right)} \tag{4.24}$$

$$E[\hat{\theta}_{\text{media}}|S_n = i] = \frac{1}{p} \cdot \frac{F_E(8, 7p, b_{i+1}) - F_E(8, 7p, b_i)}{F_{\hat{\theta}_n}(b_{i+1}) - F_{\hat{\theta}_n}(b_{i+1})} \tag{4.25}$$

As the closed-form BUP considers only two states, $i$ can have a value of 0 or 1. Consequently, $b_i$ and $b_{i+1}$ can have values as in Eq. 4.27. $F_E(H, u, k)$ is provided in Eq. 4.26. $R_i$ is defined as Eq. 4.28. Detailed derivation of $q_{ii}$ is provided in [160], and is not discussed in this thesis.

$$F_E(H, u, k) = 1 - \sum_{h=0}^{H-1} \frac{(u \cdot k)^h}{h!} \cdot e^{-u \cdot k} \tag{4.26}$$

$$\begin{cases} b_0 = 0 \\ b_1 = \hat{\theta}_{\text{media}} \\ b_2 = \infty \end{cases} \tag{4.27}$$

$$R_i \approx R\left(\frac{7p}{6} \cdot \frac{F_E(6, 7p, b_{i+1}) - F_E(6, 7p, b_i)}{F_{\hat{\theta}_n}(b_{i+1}) - F_{\hat{\theta}_n}(b_{i+1})}, rtt\right) \tag{4.28}$$

In conclusion, the closed-form BUP [160] can be presented as a function of buffer size ($x$), round trip time ($rtt$), TFRC receiving rate ($R_R$) and consuming rate of delivered multimedia ($R_S$) which is presented as Eq. 4.29.

$$
\begin{aligned}
\gamma(x) =& F(x, rtt, R_R, R_S), \\
& x_i > 0
\end{aligned}
\tag{4.29}
$$

### 4.4.4 BCM Buffer Underflow Avoidance Scheme

The proposed BCM employes a novel **buffer underflow avoidance scheme (BUAS)** which determines initial buffer estimation $S_{init}$ for play buffer used in QAMMD. $S_{init}$ can be the VBV buffer trigger ($S_{VBV}$) or assembly buffer trigger ($S_{ab}$). $S_{VBV}$ is easily calculated at encoding time [186]. In case of VBR, a verified VBV buffer size is given during encoding time. For CBR, $S_{VBV}$ can be determined with VBV delay and data rate instead of a given value. $S_{ab}$ is the initial size estimation of the assembly buffer which is chosen as buffer size estimations by ISSs. BUAS proposes an approach to estimate the initial buffer size of the assembly buffer, $S_{ab}$ using a single TFRC (BUP) analytic model [160] for each ISS. Simply, BUAS selects $S_{init}$ as the biggest value from $S_{VBV}$ and $S_{ab}$.

Following the results of section 4.4.3, The BUP of $i$, $\gamma_i(x_i)$ in closed-form make use of several parameters including round trip time($rtt$), $R_{R_i}$ and $R_{S_i}$ since TFRC uses an equation based on loss and round trip time to determine bandwidth. Consequently, $\gamma_i(x_i)$ could be described as in Eq. 4.30.

$$
\begin{aligned}
\gamma_i(x_i) =& F(x_i, rtt_i, R_{R\_i}, R_{S\_i}), \\
& x_i > 0
\end{aligned}
\tag{4.30}
$$

BUAS considers that assembly buffer underflow occurs when all ISSs have reached underflow. Using this assumption, the overall BUP ($P$) is the product of BUPs of all $n$ ISS's as in Eq. 4.31

$$
P = \prod_{i=0}^{n} \gamma_i(x_i) \tag{4.31}
$$

In order to achieve the given target BUP, $P_{target}$ in the assembly buffer, BUAS estimates the initial buffer size with Eq. 4.32. $P_{target}$ is dependent on the number of users to be supported. The higher the number of users, the smaller probability of buffer underflow is required. For example, a good target value for 200 users is 0.005. A smaller value of $P_{target}$ supports more users with a certain service level.

$$\begin{aligned} S_{ab} &= \sum_{i=0}^{n} \gamma_i^{-1}((P_{target})^{\frac{1}{n}}) \\ &= \sum_{i=0}^{n} F^{-1}((P_{target})^{\frac{1}{n}}, rtt, R_{R_i}, R_{S_i}) \end{aligned} \tag{4.32}$$

In summary, BUAS determines initial buffer size $S_{init}$ before providing frame data to the decoder using $S_{VBV}$ determined during encoding time and $S_{ab}$ which is estimated periodically using target assembly buffer underflow probability ($P_{target}$), received data rate ($R_{R_i}$), data rate to the decoder ($R_{S_i}$) and inverse function of ISS BUP function $F^{-1}(\cdot)$.

### 4.4.5 Segmented Data Streaming Scheme

In order to use data replicated with MDRS, QAMMD needs to search segments and to deliver those from the multiple-servers. As assumed in the Smart PIN design, the search function is supported in overlay network processing. However, some of the segmented content for realtime delivery (e.g. VAR_SEG) is not fully replicated to a specific node, presented in Fig. 4.15, as data replication scheduling for data delivery.

Streaming from nodes which include the whole content is the ideal case of QAMMD since the initial buffer prediction using bandwidth estimation is more precise with a fixed number of nodes during the streaming service. However, data replication is performed anytime, and it is also useful to use more nodes in order to have higher overall throughput although they do not include all parts of the content. In addition, the buffer for received data supports using partly replicated content, since there is previously received data.

There are several assumptions for rate allocation and packet partition to the source. As

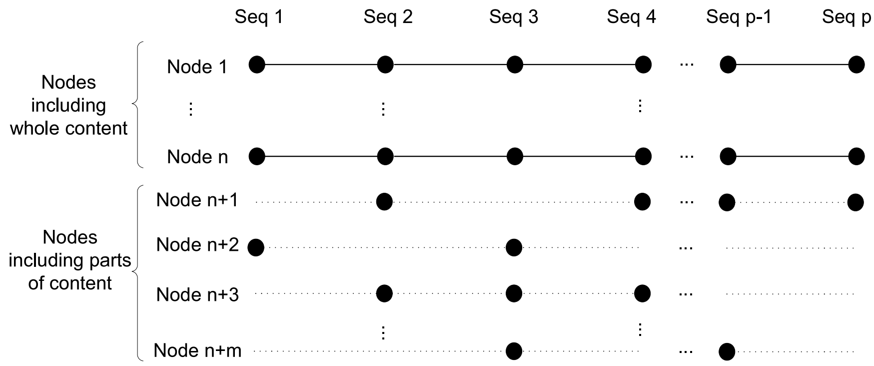



Figure 4.15: An Example of Data Replication of MDRS for Streaming

mentioned, both of them start only with nodes which have whole content. The allocated rate is proportional to the rate which the node can transmit. The packet partition can be server based approach similar to [85] or receiver-based approach similar to [156]. Smart PIN can use either way, but assumes that second one is used.

### 4.4.6 Summary

This section describes a novel **Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD)** which employs a double virtual receiving buffer architecture to maintain quality at high levels in highly loaded conditions during multimedia delivery. QAMMD also employs a buffer underflow avoidance scheme (BUAS) which best balances the flow of data between the multiple connections and a play buffer in order to achieve high multimedia quality without content adaptation to network conditions. In our discussion on streaming using segmented data, nodes including whole content will support the ideal case for buffer prediction, and nodes which include partly-replicated data are also useful in order to utilise more resources, such as bandwidth.

## 4.5 Chapter Summary

In this chapter we discuss the system architecture for the Smart PIN design from the view of a network system, we discuss the service architecture from the view of a context-aware personal information system, and we discuss the scenario analysis from the view of service. Since there are several existing approaches for the service model and system components, we focus on specific issues such as tertiary services and service component.

As a performance and cost-oriented context-aware personal information network, Smart PIN includes two novel mechanisms which are Multimedia Data Replication Scheme (MDRS) and a Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD). MDRS supports efficient user access to information located on remotely distributed devices in a heterogeneous network environment. MDRS includes data selection and data pair (content and context) delivery based on a utility function and minimum data set requirements.

QAMMD employs a double virtual receiving buffer architecture to maintain quality at high levels in highly loaded conditions during multimedia delivery through a buffer underflow avoidance scheme (BUAS) which achieves high multimedia quality without content adaptation to network conditions. QAMMD supports multiple-source streaming with nodes with whole contents and nodes with partly replicated contents. Following this chapter, the thesis includes modelling, simulation, and test results as the result of research.

# Chapter 5

# Modelling, Simulation-based Test and Results Analysis

## 5.1 Introduction

This chapter presents modelling of Smart PIN contributions, including the proposed Multimedia Data Replication Scheme (MDRS) and Quality Adaptive Multiple-source Multimedia Delivery (QAMMD) using Network Simulator 2 (NS-2)[1]. The Smart PIN simulation model is presented in Fig. 5.1, and it includes network and service components as well as applications. Models of the main contributions of Smart PIN are presented in content delivery, replication management and presentation adaptation blocks in the service component as well as a content sharing and multiple-source streaming in the application block. These are indicated with bold in Fig. 5.1.

First of all, the NS-2 simulator is introduced. NS-2 is used for both modelling and setting up simulation environment. This chapter presents some network models: an overlay model and a heterogeneous network model. In addition, data models supporting context-content pair and multimedia data for streaming are introduced. Application models are separated into a content sharing application for MDRS and a multiple-source streaming

[1]Network Simulator 2, http://www.isi.edu/nsnam/ns/, last accessed 18 Nov. 2009

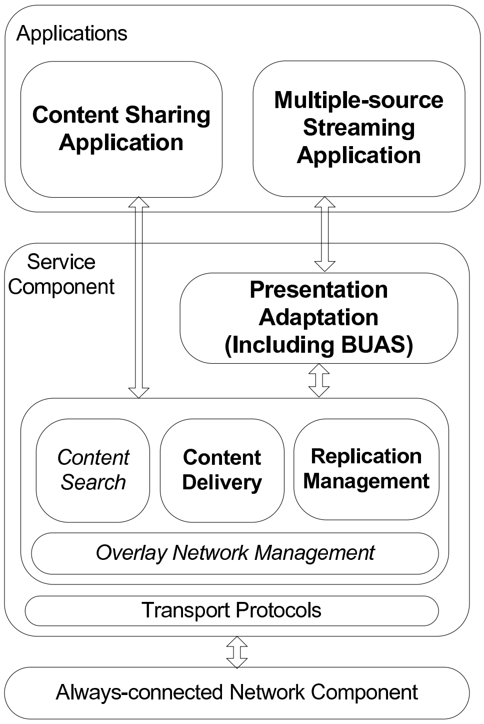



Figure 5.1: Simplified Smart PIN Architecture

application for QAMMD. These are based on NS-2 application models. In order to evaluate the performance of MDRS and QAMMD, comparisons with similar approaches are included.

Test scenarios are introduced along with network topologies which are used for the simulation tests. The test results and data analyses cover data replication and multiple-source streaming for several topologies, respectively. Each test scenario includes background traffic; content sharing applications have streaming traffic using UDP and TFRC and multiple-source streaming applications have data replication traffic delivered using TCP.

## 5.2 Simulation Models

### 5.2.1 NS-2 Simulation Models

NS-2.31 is the version used for the simulation of Smart PIN. NS-2 is a discrete event simulator which provides substantial support for the simulation of data delivery using various protocols in wired and wireless networks. NS-2 is heavily used by the network research community[2]. Since the main user interfacing tool of NS-2 is Tcl/Tk [3], the NS-2 application operates as a Tcl interpreter, the simulation scripts are based on Tcl syntax, and other applications such as the network animator (NAM), are implemented with Tk GUI widgets. For enhancement of simulation scripts, OTcl[4], an object oriented extension of Tcl, is used in NS-2. Most of object modules are implemented in the C++ language.

Fig. 5.2 presents user's prospective view of NS-2. When a user provides an OTcl script as input, NS-2 generates trace and NAM file(s) as a result. The trace file includes information on the packets which are transmitted, and received at the nodes. NAM presents packet delivery and mobility information of mobile nodes in an animation. As mentioned earlier, NS-2 has both C++ and OTcl parts. Because of the duality, the objects are implemented in C++ and OTcl separately, and they intercommunicate. However, not all of the implementation is required to have a duality relationship. If some function is efficient in C++, it is only implemented in C++ and vice versa.

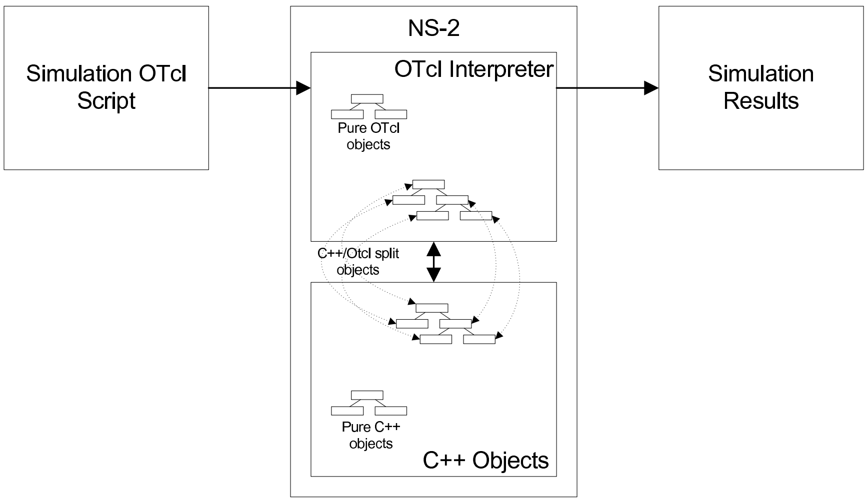



Figure 5.2: Simplified NS-2 Structure

An NS-2 simulation runs on the link layer and the layers above as shown in Fig. 5.3.

[2]ns-3 monthly report: November 2006, http://www.nsnam.org/docs/monthly/november06.html, last accessed 18 Nov. 2009

[3]Tcl Developer Site, http://www.tcl.tk/, last accessed 18 Nov. 2009

[4]OTcl: http://bmrc.berkeley.edu/research/cmt/cmtdoc/otcl/, last accessed 30 Aug. 2009

Physical models are adopted in the channel object. **Nodes** represent computer network nodes which support various protocols and applications. An ***agent*** models a transport layer protocol such as UDP, TCP, TFRC, etc. There can be more than one agent for a node. An agent can support an Application (App) which means a service which make use of the protocol. Some agents do not require to have an App. More information about NS-2 is available in the NS documentation[5].

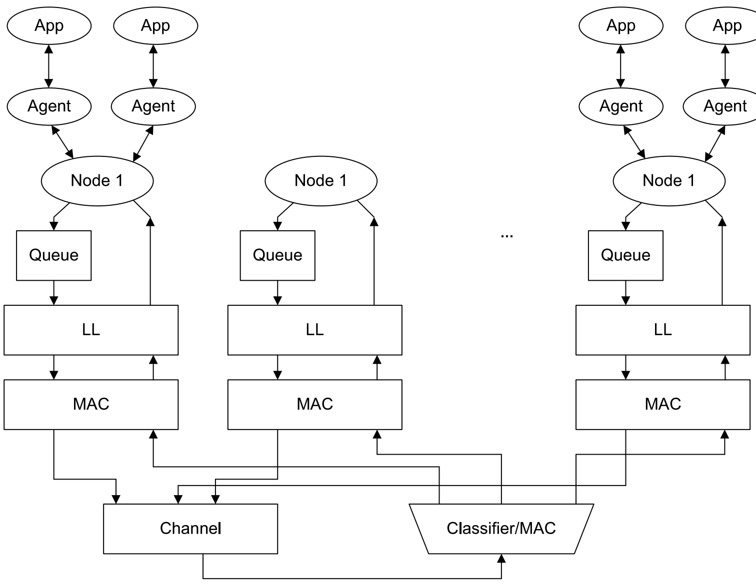



Figure 5.3: An Example of NS-2 Layer Architecture based on LAN

### 5.2.2 Network Technologies

#### 5.2.2.1 Wired Network Model

The wired network model is a basic network model which is used in NS-2. The model includes links and queues between the nodes. Each link is simplex which supports one direction for the link. However, there is also a way to setup a duplex link which involves the creation of a simplex link for each direction. Each link needs to have bandwidth and delay as its parameters. In addition, the type of queue should also be defined, such as

[5]The Network Simulator ns-2: documentation, http://www.isi.edu/nsnam/ns/ns-documentation.html, last accessed 18 Nov. 2009

DropTail, which drops packet from the queue's tail when queue overflow happens.

When wired and wireless networks are used together (i.e. wired-cum-wireless scenario), hierarchical addressing should be used in NS-2. With one network technology such as wireless, the node address could be assigned automatically. However, the address of each node should be allocated manually and different technologies need to have a separate domain of address space, when multiple technologies are used. An example for hierarchical addressing is presented in section 5.2.2.5.

#### 5.2.2.2 WLAN Model

IEEE 802.11g WLAN was used for simulation based on the NS-2 default implementation. The detailed parameters for NS-2 IEEE 802.11g model for physical layer and MAC layer are presented in Table 5.1 and 5.2. Depending on the coverage of WLAN, the values of Pt_, RXThresh_ and CSThresh_ can be changed. The Destination-Sequenced Distance Vector routing (DSDV), Ad-hoc On-Demand Distance Vector Routing (AODV) are used for ad-hoc routing. NO Ad-Hoc (NOAH)[6] is also used in order to simulate the realistic environment which is adopted for conventional WLAN access points.

| Parameter | Value |
|---|---|
| Freq_ | 2.4GHz |
| Pt_ | $2.81838 \times 10^{-1}$ |
| RXThresh_ | $7.74636 \times 10^{-9}$ |
| CSThresh_ | $7.74636 \times 10^{-9}$ |

Table 5.1: NS-2 IEEE 802.11g physical layer parameters

In Table 5.1, Feq_ indicates what frequency is used. Pt_ is the transmit power of a node. RXThresh_ represents signal strength of one frame received by the receiver. CSThresh_ is the carrier sense threshold used to determine whether one frame is detected by the receiver.

In Table 5.2. SlotTime_ is a unit of back-off delay. SIFS_ represents the short interframe space. PreableLength_ means the length of the Physical Layer Convergence Procedure

[6]NO Ad-Hoc Routing Agent (NOAH): http://icapeople.epfl.ch/widmer/uwb/ns-2/noah/, last accessed 18 Nov. 2009

| Parameter | Value |
|---|---|
| SlotTime_ | 9usec |
| SIFS_ | 16usec |
| PreambleLength_ | 96 bits |
| PLCPHeaderLength_ | 40 bits |
| dataRate_ | 54 Mbps |
| basicRate_ | 6 Mbps |

Table 5.2: NS-2 IEEE 802.11g MAC layer parameters

(PLCP) preamble. PLCPHeaderLength_ is the length of PLCP message header. DataRate_ is the rate for data frames, and finally, BasicRate_ is the rate for the control frames.

#### 5.2.2.3 WiMAX Model

The NIST IEEE 802.16 module[7] is used as the WiMAX extension for NS-2 in this thesis. This is based on the IEEE 802.16 standard [18] and the mobility extension 80216e-2005 [188]. This model is extended as a subclass of the NS-2 802.11 model including the physical layer (Phy/WirelessPhy/OFDM) and MAC (e.g. Mac/802_16). As the MAC operation of IEEE 802.16 is different from that of 802.11, MAC configuration is required before the simulation starts, including address classifier (i.e. SDUClassifier), MAC interfacing (i.e. WimaxScheduler), channel, etc. A subscriber station and a base station have different features in WimaxScheduler, so they are implemented as separate classes (i.e. WimaxScheduler/SS and WimaxScheduler/BS). The detailed parameters for NS-2 IEEE 802.16 model for physical layer and MAC layer are presented in Table 5.3 and 5.4.

| Parameter | Value |
|---|---|
| Freq_ | 2.4GHz |
| Pt_ | 0.025 |
| RXThresh_ | $1.26562 \times 10^{-13}$ |
| CSThresh_ | $1.012496 \times 10^{-13}$ |

Table 5.3: NS-2 IEEE 802.16 physical layer parameters

Table 5.3 includes the same parameters which are used in Table 5.1. In Table 5.4,

[7]EMNTG Seamless and Secure Mobility: http://w3.antd.nist.gov/seamlessandsecure/download.html, last accessed 18 Nov. 2009

| Parameter | Value |
|---|---|
| dcd_interval_ | 5 secs |
| ucd_interval_ | 5 secs |
| Default modulation | OFDM_16QAM_3_4 |
| t21_timeout_ | 0.02 secs |
| client_timeout_ | 50 secs |

Table 5.4: NS-2 IEEE 802.16 MAC layer parameters

dcd_inteveral_ is Downlink Channel Descriptor message interval. ucd_interval_ represents Uplink Channel Descriptor message interval. Default_modulation means the modulation method which is used. T21_timeout_ is DL-MAP message (WiMAX management message) timeout value. Client_timeout_ is a timer value for removing clients which do not communicate with the base station.

#### 5.2.2.4 Conventional Broadband Model

Residential broadband [8] NS-2 extension is also used for modelling the **wired broadband access network** as it supports asymmetric links and enables us to model broadband connectivity such as cable and DSL networks. The network parameters for the broadband links between nodes used in the simulation are shown in Table 5.5. Similar to conventional broadband, downstream and upstream links have different bandwidths: 4 Mbps and 220 kbps respectively.

| Parameter | Downstream | Upstream |
|---|---|---|
| delayMicros | 5 usecs | 0 usecs |
| capacityPackets | -1 | -1 |
| capacityBytes | 60000 bytes | 60000 bytes |
| lineRateBps | 4 Mbps | 220 kbps |

Table 5.5: Parameters for broadband network model

In Table 5.5, delayMicros represents the delay value in a broadband connection. capacityPackets and capacityBytes indicate the maximum buffer size in terms of packets and bytes. Only one of these two parameters can be used, whereas lineRateBps is the data rate

[8] Characterizing Residential Broadband Networks, http://broadband.mpi-sws.mpg.de/residential/, last accessed 18 Nov. 2009

of a broadband connection.

#### 5.2.2.5 MIH Model and Multiple Interface Support

The NIST Mobility extension[9] supports IEEE 802.21 MIH based on draft 3 of IEEE 802.21 specification [189] using UMTS, WiMAX, WLAN and Bluetooth technologies. NS-2 does not support multiple interfaces in its node design as only one physical layer is supported at a time. The NIST Mobility package includes multiple interface node support (MultiFace Node) using a virtual node linking nodes of similar or different networking technologies as shown in Fig. 5.4. Each node needs to have a Neighbour Discovery (ND) agent in order to detect layer 3 movement and to notify the MIH User agent (i.e. interface manager), which is located in a MultiFace node. The MIH User agent varies depending on user preference or network policies, therefore the MIH Function provides abstraction in order to help the implementation of the MIH User agent.

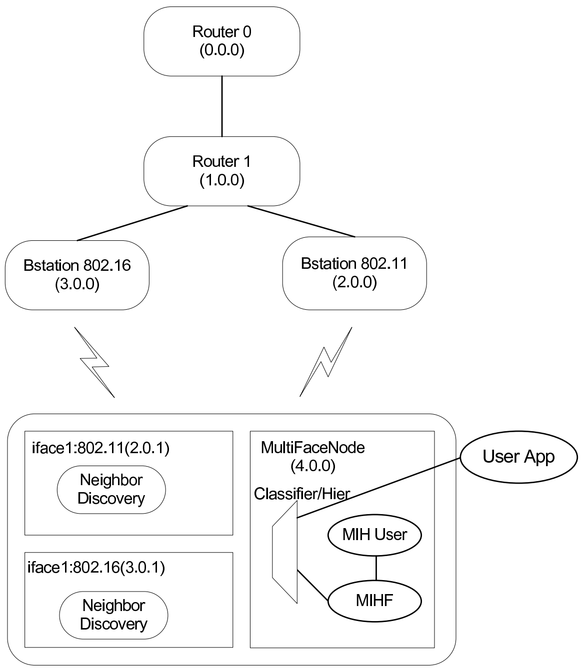



Figure 5.4: MIH Hierarchical Addressing Example with WLAN and WiMAX

[9]EMNTG Seamless and Secure Handoevers: http://w3.antd.nist.gov/seamlessandsecure/download.html, last accessed 18 Nov. 2009

### 5.2.3 Overlay Network Models

As indicated in the design section, Smart PIN is based on P2P networks. This overlay networking is not supported in the NS-2 basic package. The current overlay network model for Smart PIN is based on BitTorrentSim[10]. The original BitTorrentSim provides a BitTorrent application, a BitTorrent tracker, a BitTorrent connection and BitTorrent messages in the package. However, not all of the models for BitTorrent are required and some of them have been extended for a specific application of Smart PIN such as data replication and multiple-source streaming. Currently, the BitTorrent connection supporting TCP, TFRC and UDP and the BitTorrent tracker supporting the search function are extended for Smart PIN model.

Fig. 5.5 presents a simplified diagram of the overlay network components for Smart PIN. P2PApp is based on the NS-2 application. Similar to a BitTorrent connection, P2PApp supports the TCP protocol for data replication. In order to support multiple-source streaming, UDP and TFRC protocols are used in P2PApp. Therefore, inherited applications implemented for each scheme can use all protocols for the overlay network connection to other peers. The connection between peers has to be set up manually. P2PGlobalIndex supports the search function of overlay network as an extension of a BitTorrent tracker. Currently, network overhead for search is not considered in the simulation scenario.

### 5.2.4 Application models

#### 5.2.4.1 Simulation Data Models

Applications in the simulation tests focus on the delivery of the context and content pairs. As discussed in chapter 4.3, content includes NO_SEG, VAR_SEG and FIX_SEG data. For data context, the assumed size was uniformly distributed between 1 and 10 Kbytes. As no real measurement analysis of annotated metadata was available, it was assumed that the size of metadata is similar to that of regular web pages. The model used during simulations is

[10]Simulation of BitTorrent Peer-to-Peer (P2P) Networks in ns-2: http://www.tu-harburg.de/et6/research/bittorrentsim/index.html, last accessed 18 Nov. 2009

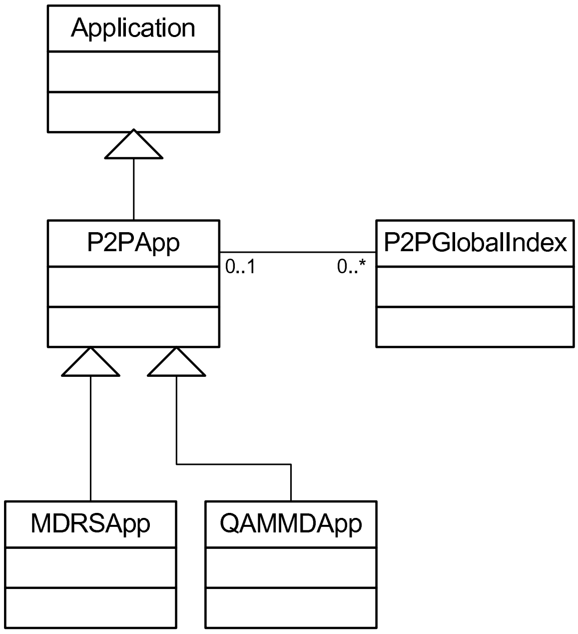



Figure 5.5: Overlay network models for simulation

adopted from the modelling of web content [62].

The generation of NO_SEG content used a size uniformly distributed between 400 Kbytes to 3 Mbytes which is consistent with that of the size distribution of still images taken with a 5M pixel digital camera. The relevance value of each data for users and the popularity value for variable length segmented data are both generated with uniform distribution and have values between 0 and 1.

5 five-minute long VBR encoded video sequences were selected from movies with different degrees of motion content: *"Die Hard 1"* with very high motion content, *"Jurassic Park 3"* with an average - high motion content, *"Don't Say A Word"* with average - low motion content, *"Family Man"* with very little action and *"The Road To El Dorado"* with average - high motion but including cartoons. Assuming broadcast and studio quality is used [158], the clips were MPEG-2 encoded at 3 Mbps and 4 Mbps using the same frame rate (25 frames/sec) and the same IBBP frame pattern (12 frames/GOP). Traces were collected from these clips and used during simulations.

The simulations use VAR_SEG multimedia sequences from the "*Die Hard 1*" movie encoded at a high quality for data replication. The movie is divided into 102 segments and each segment includes up to 5 GOPs. Data context for each segment is generated with the same assumption made for NO_SEG data metadata. The average size of the segments is 1.47 Mbytes. The minimum and maximum segment sizes are 164 kbytes and 3.42 Mbytes, respectively. These values were set in order to test the case with a wide range of content of variable size.

As shown in Table 5.6, the simulations assumes that $w_{1,T}$ and $w_{2,T}$ in the utility function (see Eq. 4.3) are 1 and 0 for private NO_SEG data and 0 and 1 for shared VAR_SEG data, respectively. FIX_SEG data are not used in the simulation.

| Weight | Private NO_SEG | Shared VAR_SEG |
|---|---|---|
| $w_{1,T}$ | 1 | 0 |
| $w_{2,T}$ | 0 | 1 |

Table 5.6: Examples of weights of utility function in simulations

#### 5.2.4.2 Data Replication Application Models

The proposed ***Multimedia Data Replication Scheme (MDRS)*** is based on the Smart PIN data replication algorithm described in chapter 4.3 and implemented as an NS-2 application as shown in Fig 5.6. As discussed in previous sections, other service component features are shared except content delivery, replication management and content sharing applications since they are related to the contributions of this thesis. Every data replication approach assumes that the TCP protocol is used as transport protocol, because it provides reliable data delivery.

Each MDRS node performs data categorisation and data delivery using the utility function and a multimedia data set availability enhancement with the minimum data set requirement, $G^k$ (see section 4.3.4). To achieve the target minimum data set, each node checks its storage and global data set status sequentially. If there is multimedia content which does not have replicated enough data segments, the replicating node picks up a node from the User Group and checks whether it can include specific segments or not.

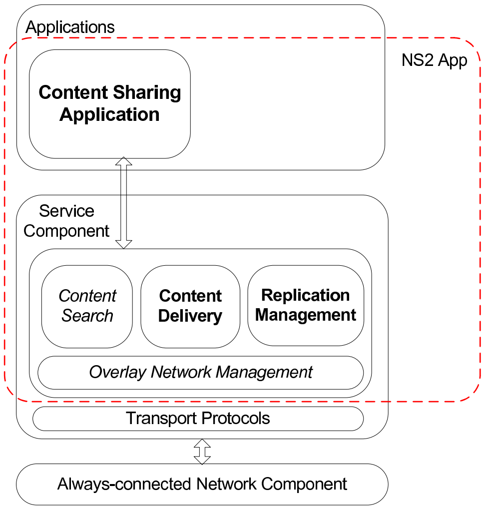



Figure 5.6: Smart PIN data replication application for simulation tests

The proposed MDRS is compared with two general purpose data replication schemes which are also modelled in NS-2: ***Autonomous Replication (AR)*** [123] and ***Tempo*** [126]. The implemented AR scheme periodically estimates data segment availability. If the estimation does not reach the target data availability, the replicating node picks up a node at random and transfers the data. Since Tempo considers its network usage, the implemented model picks up a random segment and transfers data based on its network budget (e.g. target bandwidth, see Table 5.7). The storage size of each mobile device was assumed to be 2 GBytes.

MDRS uses specific parameters which are shown in Table 5.7. The utility has a value between 0 and 1. The utility thresholds allow delivering the amount of data items which have a higher utility value than the threshold. The higher utility threshold provide, the lower number of data items are involved. In this thesis, only 0.75 utility threshold is considered for transfer and selection utility threshold. In order to achieve a certain target bandwidth, MDRS and Tempo use the same value of 1.86 Mbps. Device availability is usually very low for data sharing, around 20% [190]. However, there are also IP aliasing issues because the same node has different IP addresses [191]. Based on these, device availability for the simulations is assumed 40% in this thesis.

| | Parameter | Value |
|---|---|---|
| MDRS | Transfer utility threshold | 0.75 |
| | Selection utility threshold | 0.75 |
| | Target bandwidth ($B_{BW}$) | 1.86 Mbps |
| | Device availability | 0.4 |
| AR | Replication Interval | 10 secs |
| | Device availability | 0.4 |
| | Target data availability | 0.8 |
| Tempo | Target bandwidth ($B_{BW}$) | 1.86 Mbps |

Table 5.7: Parameters for replication schemes

In order to investigate the relationship between data replication and multimedia streaming, a server and a client are used for simulation streaming purposes during data replication. The nodes used for streaming are not involved in data replication in order to simplify the simulation and focus on one aspect at a time. The detailed description for streaming nodes will be discussed in each test scenario, respectively.

#### 5.2.4.3 Multiple-source Multimedia Streaming Models

The multiple-source streaming application is also implemented as an NS-2 application as depicted in Fig 5.7. Similar to MDRS, service component features are shared except content delivery, and presentation adaptation. In all multiple-source streaming applications, the receiver uses a buffer to collect data before feeding it to the decoder and schedules the packets for sending from the senders. Specifically, presentation adaptation includes buffer prediction algorithms such as our proposed ***Quality Adaptive Multiple-source Multimedia Delivery (QAMMD)*** and the Predictive Buffering Algorithm (PBA) [161].

In order to analyse the relationship between streaming and data replication traffic, the simulation adopts streaming approaches based on TFRC and UDP protocols which are commonly used for multimedia streaming [85, 143]. While the TFRC traffic is friendly towards TCP transfers, the UDP-based one uses the bandwidth in a greedy manner.

Modelling and simulations employ models for QAMMD, PBA, a MSDVS-like [85] multiple TFRC connections-based approach (mTFRC) and a PROMISE-like [143] UDP-

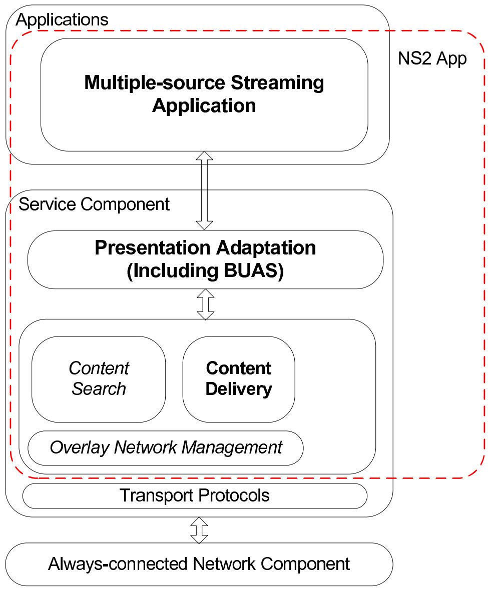



Figure 5.7: Smart PIN multiple-source streaming application for simulation tests

based multiple streaming solution (mUDP). QAMMD and PBA adopt the buffer estimation algorithm but they use different solutions. Fig 5.8 presents the flow chart of PBA. PBA uses a statistical approach which assumes the connections are independent. However, in a practical scenario, the connections use the same network resources such as Ethernet cards and might affect each other. mTFRC and mUDP do not use buffer prediction. mTFRC uses adaptive data delivery based on the TFRC protocol. mUDP uses equalised bandwidth allocation at the start of the streaming instead of dynamic bandwidth allocation which is used for the other solutions. In all approaches, the receiver requests the same packets to be delivered from the multiple senders. In addition, in these simulations all approaches adopt static peer selection and initially connect to three nodes only.

In QAMMD, $P_{target}$ is set to 0.005 and $S_{VBV}$ is set to 224 Kbytes which is determined at encoding time. The same $S_{VBV}$ is used by mTFRC and mUDP, too. All approaches use 3.2 Mbps as target bandwidth which is higher than the encoding rate of 3 Mbps in order to cover network delivery overhead.

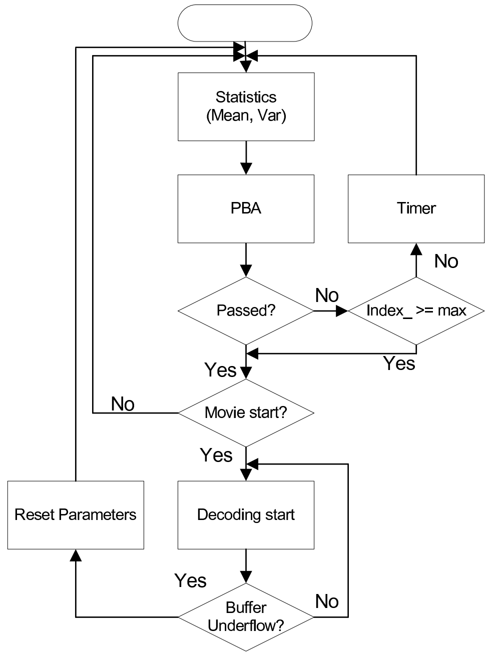



Figure 5.8: PBA flow chart

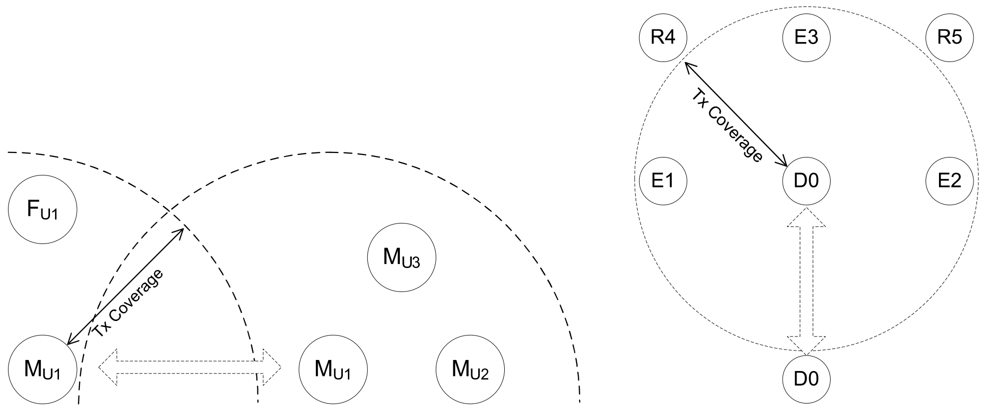
## 5.3 Network Topology and Simulation Scenarios

### 5.3.1 Wireless Only Topology with Ad-Hoc Routing

Although there are wired-only devices such as personal servers, as mentioned previously, most of the Smart PIN devices are assumed to support wireless connectivity. In this case, mobile devices are considered not to require any infrastructure, but they use ad-hoc routing protocols such as DSDV for communication. Based on this assumption, the simulation for data sharing mostly focuses on information overload situations and segmented data sharing dominated by minimum data set requirements. Due to the limited size and low effect on network performance, no metadata-only replication is considered in this thesis. In addition, ad-hoc topology is not considered when multimedia streaming is performed.

For information overload, a **simple wireless ad-hoc topology** is used for the simulations as depicted in Fig. 5.9(a) and it involves 4 mobile nodes and 3 users. User 1 controls two devices ($F_{U1}$ and $M_{U1}$) and the other users have a single mobile device each ($M_{U2}$ and $M_{U3}$). The test scenario includes movement of user 1's mobile device into the neighbourhood of the other users. Peer-to-peer communication protocols supporting node join and leave, data query and response are simulated in a simplified manner. The focus of the simulation is on Smart PIN data selection and transfer.

A **grid-like ad-hoc topology** is used for minimum data set (i.e. $G_k$ enhancement which is mentioned section 4.3.4) simulations with a simple mobility scenario as depicted in Fig. 5.9(b). In total, 6 mobile nodes are involved, numbered from 0 to 5. The data node (labelled as D0) includes movie fragments as segmented data (VAR_SEG), and some non-segmented data (NO_SEG). Empty nodes (marked as E1, E2 and E3) initially do not store any data. Because of communication range limitations, E1, E2 and E3 cannot transfer data directly between themselves. However, connecting nodes R4 and R5 enable the connectivity between the Smart PIN nodes (i.e. E1, E2 and E3) when D0 leaves away.



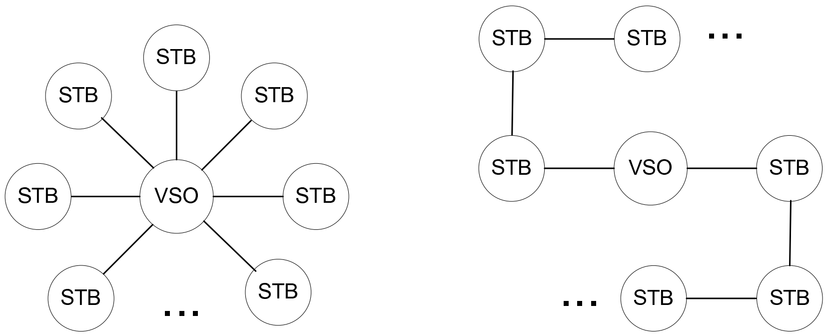

(a) Simple ad-hoc topology (b) Grid-like topology

Figure 5.9: Ad-hoc network simulation topologies

The test scenario for grid-like topology includes periodic movement of node D0 in and out of the network range (i.e. trips). The simulation is assessed with different numbers of such trips: 20, 40, 80 and 120. Approximately 20 trips take an hour to complete. The data node has a device availability of 0.4 due to the movements whereas the other devices have device availability of 1 in this scenario. A comparison based assessment includes several approaches such as Smart PIN without minimum data set (**SPIN**), Smart PIN without $G_k$ enhancement (**SPIN-NG**), Autonomous Replication (**AR**) [123] and **Tempo** [126].

### 5.3.2 Conventional Broadband Networks

As shown in Fig. 5.10, two network topologies for conventional broadband networks are considered: a star topology used for a DSL access network and a string topology modelling a cable network [192, 193]. The simulation assumes that the User Group includes 20 set-top boxes (STBs) and 1 video service office (VSO), therefore, a total of 21 simulation nodes are involved. The delivery of multimedia segments is performed using the TCP protocol which is implemented in NS-2.

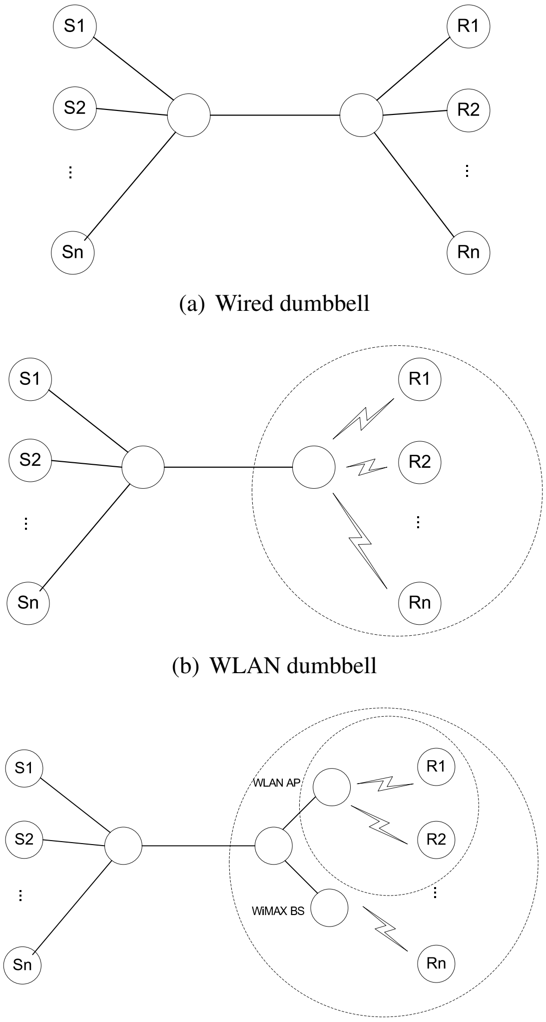



(a) Star topology (b) String topology

Figure 5.10: Network topology used for simulation (4 Mbps downlink, 220 kbps uplink)

### 5.3.3 Conventional Internet: Dumbbell Topologies

The "**dumbbell**" topology (shown in Fig. 5.11) is a typical model of the Internet [194] and is a popular topology for streaming applications [195, 196, 197]. The dumbbell topologies used in the simulations have a bottleneck with 200 Mbps bandwidth and 5 msecs delay in the middle. The other links have even the same parameters as the bottleneck link, so there is no significant drop of packets. All the queues between the nodes are drop-tail queues which have a limit of 2000 packets. However, wireless links are actual bottlenecks since they have less bandwidth than the bottleneck link in Fig 5.11(b) and Fig 5.11(c). Based on simple movement of mobile nodes, MIH is used for handover as illustrated in Fig 5.11(c). As mentioned, these topologies are used for multiple-source streaming. Data replication traffic is emulated with background TCP traffic.




(a) Wired dumbbell

(b) WLAN dumbbell

(c) WLAN-WiMAX dumbbell

Figure 5.11: Dumbbell simulation topology

## 5.4 Test Results and Analysis

### 5.4.1 Metrics for Simulation Test Assessment

The metrics for comparison are data availability and network usage. *Data availability* is measured with the average online rate of the device in the range of communication and the ratio of data residing duration over total test time on a specific device as described in Eq. 5.1 and Eq. 5.2 where $s^k_{l,j}$ is the availability of the segment $l$ of the multimedia data item $k$ in a device $j$, $P_j$ is the availability of a device, and total number of segments of the multimedia data $k$ ($L$) and total number of devices ($J$). *Network resource usage* is measured by the rate of data received by each device. Loss is measured by the rate of data dropped in all devices.

$$R_{l,j} = \frac{Data\ l\ residing\ time\ in\ device\ j}{Total\ test\ duration} \tag{5.1}$$

$$D^k_{avg} = \frac{\sum_{l=1}^{L}\sum_{j=1}^{J}(s^k_{l,j} \cdot P_j \cdot R_{l,j})}{L \cdot J} \tag{5.2}$$

In this chapter, Peak Signal to Noise Ratio (PSNR) is estimated according to the formula presented in Eq. 5.3 as multimedia quality is influenced by the delivery through the communication channel [198]. In Eq. 5.3, $MAX\_Bitrate$ is the average bitrate of the multimedia stream after the encoding process, $EXP\_Thr$ is the average throughput expected when adaptively delivering the multimedia stream over the network and $CRT\_Thr$ is the actual throughput measured during delivery. The detailed derivation of Eq. 5.3 is presented in Appendix A.

$$PSNR = 20 \cdot log_{10}\left(\frac{MAX\_Bitrate}{\sqrt{(EXP\_Thr - CRT\_Thr)^2}}\right) \tag{5.3}$$

In order to provide strong statistics of simulation tests, various approaches are used. Data replication tests use long simulation time (up to 2 hours), different number of data items. Multiple-source streaming tests use many nodes which have different kinds of movies and different starting points of those. If there could not be many nodes, test includes several runs (up to 10 times for each test).

### 5.4.2 Data Replication Test Results and Analysis

#### 5.4.2.1 Tests with Wireless Ad-Hoc Networks

Two topologies are considered in this section: the **simple wireless ad-hoc topology** with 5 nodes and the **grid-like ad-hoc topology** with 6 nodes. The simple wireless ad-hoc topology is used for information overload situation and adopts a simple mobility scenario of a node as mentioned in subsection 5.3.1. The comparison is done with AR and Tempo schemes implemented with NS-2. All models used employ no data segmentation.

Fig. 5.12 illustrates network usages when each scheme was employed in turn using the DSDV ad-hoc routing protocol. Since the actual target changes according to the utility function, Smart PIN uses an average of 63 kbps, whereas Tempo uses an average of 156 kbps. The AR scheme is based on a periodic time and network usage which depends on the amount of data transferred in that time. The AR scheme uses on average 494 kbps in terms of network bandwidth.

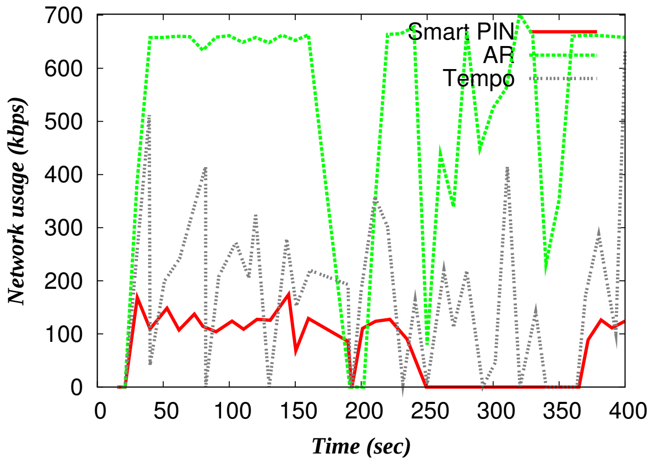



Figure 5.12: Network usage graph with simple wireless topology using DSDV

Table 5.8 and Fig 5.13 present data availability against the total number of data items stored on devices. With the assumption that device availability is similar, data availability could be measured as the duration for which data existed on the mobile devices over the total test time (i.e. about an hour). Smart PIN shows much higher data availability than the other schemes, regardless of the number of items. In particular it is very significant to note that when there are 200 data items, Smart PIN data availability is 59% higher than that of AR and 76% higher than that of Tempo. For 1500 data items, Smart PIN achieves 39% and 73% higher data availability than that of AR and Tempo, respectively.

Fig. 5.14 illustrates network usage when each scheme was employed in turn using AODV ad-hoc routing protocol. Similar to DSDV tests, Smart PIN uses on average 64

| Data item | Smart PIN | | AR | | Tempo | |
|---|---|---|---|---|---|---|
| | Avg. | Std. Dev. | Avg. | Std. Dev. | Avg. | Std. Dev. |
| 200 | 0.97 | 0.02 | 0.61 | 0.27 | 0.55 | 0.29 |
| 500 | 0.92 | 0.06 | 0.55 | 0.29 | 0.50 | 0.29 |
| 1000 | 0.82 | 0.13 | 0.52 | 0.28 | 0.50 | 0.29 |
| 1500 | 0.71 | 0.20 | 0.51 | 0.28 | 0.49 | 0.28 |

Table 5.8: Data availability statistics with simple wireless topology using DSDV

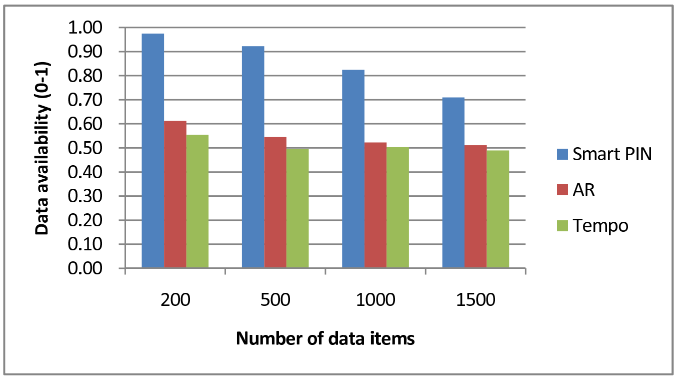



Figure 5.13: Data availability chart with simple wireless topology using DSDV

kbps, whereas Tempo uses on average 152 kbps. As the AR scheme is based on a periodic time, network usage depends on the amount of data which is transferred in the duration. The AR scheme uses on average 509 kbps in terms of network bandwidth.

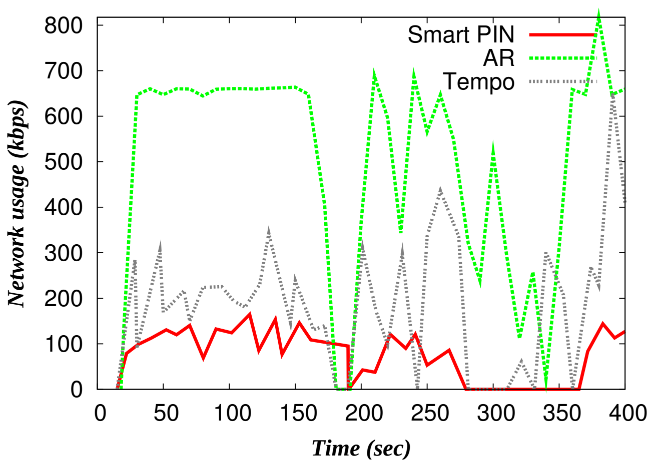



Figure 5.14: Network usage graph with simple wireless topology using AODV

Table 5.9 and Fig 5.15 present data availability against the total number of data items stored on devices. The test results based on AODV are similar to the test results based on DSDV. Smart PIN shows a much higher data availability than the other schemes, regardless of the number of items. In particular it is very significant to note that when there are 200 data items, Smart PIN shows 57% and 92% higher data availability than that of AR and Tempo, respectively. For 1500 data items, Smart PIN achieves data availability with 38% higher than AR and with 30% higher than Tempo.

| Data item | Smart PIN | | AR | | Tempo | |
|---|---|---|---|---|---|---|
| | Avg. | Std. Dev. | Avg. | Std. Dev. | Avg. | Std. Dev. |
| 200 | 0.96 | 0.02 | 0.61 | 0.29 | 0.50 | 0.28 |
| 500 | 0.92 | 0.06 | 0.55 | 0.28 | 0.50 | 0.29 |
| 1000 | 0.82 | 0.13 | 0.52 | 0.28 | 0.52 | 0.29 |
| 1500 | 0.69 | 0.22 | 0.50 | 0.28 | 0.53 | 0.29 |

Table 5.9: Data availability statistics with simple wireless topology using AODV

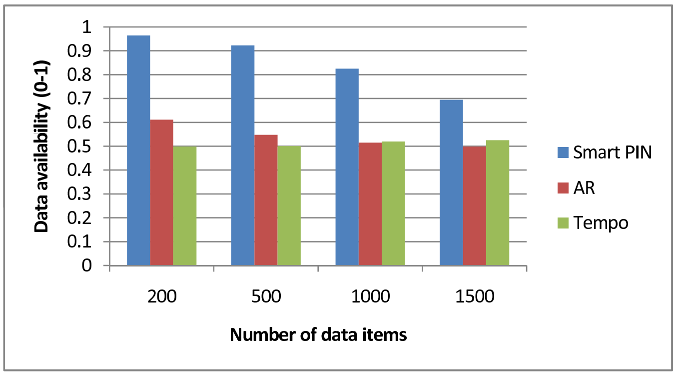



Figure 5.15: Data availability chart with simple wireless topology using AODV

The **grid-like ad-hoc topology** is mainly used to see how minimum data set requirement affects the metrics related to data replication. As shown in Table 5.10, Smart PIN without consideration of $G_k$ (SPIN-NG) shows superior data availability to other approaches such as AR and Tempo since it limits replicated data based on utility in the view of overall data. However, it does not consider the minimum data sets of VAR_SEG data to be replicated in order to support wireless P2P data streaming. Smart PIN with $G_k = 3$ (SPIN) in this test scenario shows only small differences in the overall data availability and better data availability of the VAR_SEG segments. Especially, VAR_SEG segment availability increases with the increase in the number of trips as shown in Fig. 5.16. Due to the movement of the data node, data availability in the overall system is quite low. However, data availability for VAR_SEG data is 0.94 and the overall data availability is 0.78 in the nodes which are empty and not moving. Considering the total replicated sets of multimedia data segments, Smart PIN achieves 2.56 sets stored on average across devices when the target for ideal data availability was $G_k = 3$, representing the number of sets of segments to be replicated.

| | Total data | | NO_SEG data | | VAR_SEG data | | |
|---|---|---|---|---|---|---|---|
| | Num | Avg. DA | Num | Avg. DA | Num | Avg. DA | Achieved $G_{avg}^{L,k}$ |
| SPIN | 2729 | 0.78 | 2468 | 0.74 | 261 | 0.94 | 2.56 |
| SPIN-NG | 2634 | 0.82 | 2468 | 0.82 | 166 | 0.80 | 1.63 |
| AR | 4259 | 0.64 | 4042 | 0.64 | 217 | 0.64 | 2.13 |
| Tempo | 2946 | 0.51 | 2780 | 0.51 | 166 | 0.52 | 1.63 |

Table 5.10: Data availability (DA) when having 2000 data items in the system with grid-like topology using DSDV

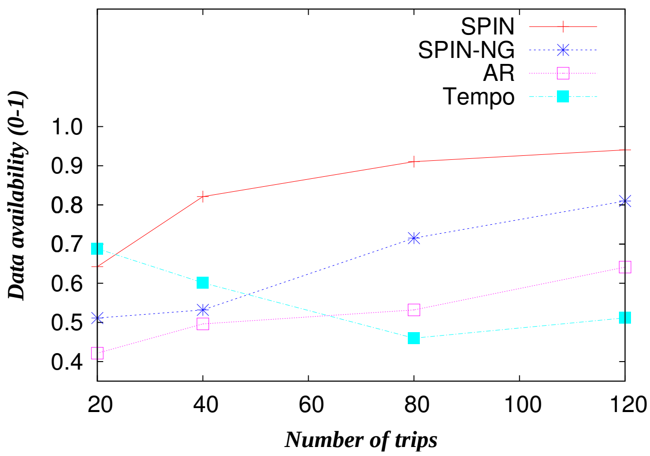



Figure 5.16: VAR_SEG data availability with grid-like topology using DSDV

Fig. 5.17 illustrates network load for the different schemes. Since the actual target bandwidth also changes according to the utility function, Smart PIN uses 36 Kbps on average, whereas Tempo uses an average of 51 Kbps. Since the AR scheme is based on a periodic time (i.e. 10 sec. is the inter-replication interval), network load depends on the

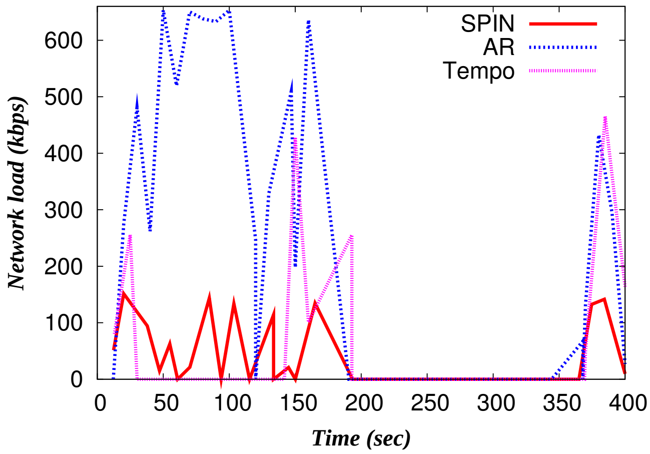



Figure 5.17: Network usage graph with grid-like topology using DSDV

amount of data which is transferred in that time. The AR scheme uses on average 195 Kbps network bandwidth.

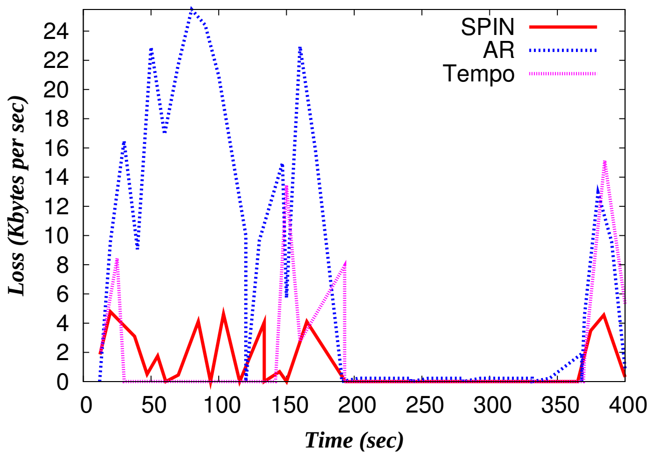



Figure 5.18: Data loss graph with grid-like topology using DSDV

Fig. 5.18 illustrates data loss for each scheme during the simulation. Smart PIN shows a 2.5% loss of data on average, which is similar to the results for AR and Tempo under the same conditions. However, as Fig. 5.18 shows the burstiness of loss is much higher for the two schemes in comparison with Smart PIN.

Similar to the DSDV test, Table 5.11 shows that Smart PIN without consideration of $G_k$ (SPIN-NG) has superior data availability in comparison to the other approaches using the AODV routing protocol. Smart PIN with consideration of $G_k$ (SPIN-NG) shows higher data availability for VAR_SEG data. In addition, VAR_SEG segment availability increases with the increase in the number of trips as shown in Fig. 5.19. AR shows a bigger number of data sets. AR achieves 2.67 sets and SPIN achieves 2.63 sets. However, more bandwidth should be used in order to achieve a higher total data set number which does in comparison with Smart PIN.

| | Total data | | NO_SEG data | | VAR_SEG data | | |
|---|---|---|---|---|---|---|---|
| | Num | Avg. DA | Num | Avg. DA | Num | Avg. DA | Achieved $G_{avg}^{L,k}$ |
| SPIN | 2973 | 0.82 | 2705 | 0.79 | 268 | 0.96 | 2.63 |
| SPIN-NG | 3057 | 0.83 | 2835 | 0.83 | 223 | 0.81 | 2.19 |
| AR | 5185 | 0.57 | 4913 | 0.57 | 272 | 0.58 | 2.67 |
| Tempo | 3515 | 0.51 | 3332 | 0.51 | 183 | 0.52 | 1.79 |

Table 5.11: Data availability (DA) when having 2000 data items in the system with grid-like topology using AODV

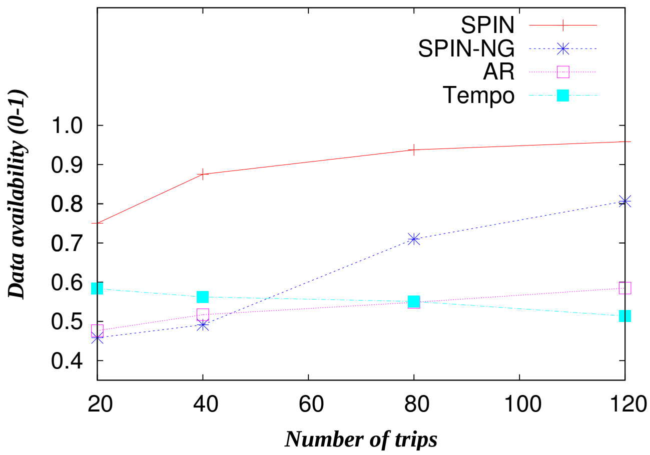



Figure 5.19: VAR_SEG data availability with grid-like topology using AODV

Fig. 5.20 illustrates network load when each scheme was employed in turn. Since the actual target bandwidth also changes according to the utility function, Smart PIN uses 105 Kbps on average, whereas Tempo uses an average of 135 Kbps. The AR scheme uses on average 282 Kbps in terms of network bandwidth.

Fig. 5.21 illustrates data loss for each scheme during the simulation. Smart PIN shows

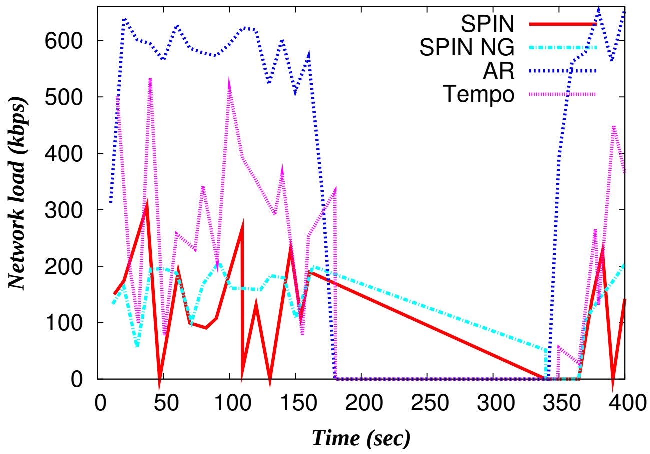



Figure 5.20: Network usage graph with grid-like topology using AODV

a 1.75% loss of data on average, whereas AR and Tempo show a loss of 2.28% and 1.65%, respectively. In addition, Fig. 5.21 shows that the burstiness of loss is much higher for the other schemes in comparison with Smart PIN.

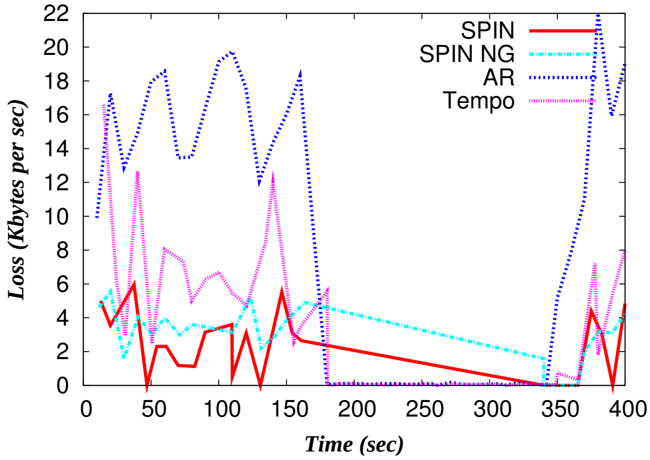



Figure 5.21: Data loss graph with grid-like topology using AODV

#### 5.4.2.2 Tests with Conventional Broadband Network

As discussed in section 5.3.2, a conventional broadband network is also used for the simulations. This test case is relevant to IPTV applications in particular. Test scenarios include recording, data replication and streaming as depicted in Fig. 5.22. The dots represent segments of recorded multimedia data. The recording operation uses the multimedia sequences provided. When a segment is generated, recording nodes store it in their own storage space. When the recording starts, each data replication scheme is used in turn for comparison. Depending on the scheme used, replicated nodes include portions of data segments from recording nodes. Since a number of recording nodes are involved, the initial number of data segments is dependent on that. Each scheme is involved with a different number of recording nodes for different test scenarios. Based on the topology presented in Fig. 5.23, streaming simulation lasts 200 seconds in each test.

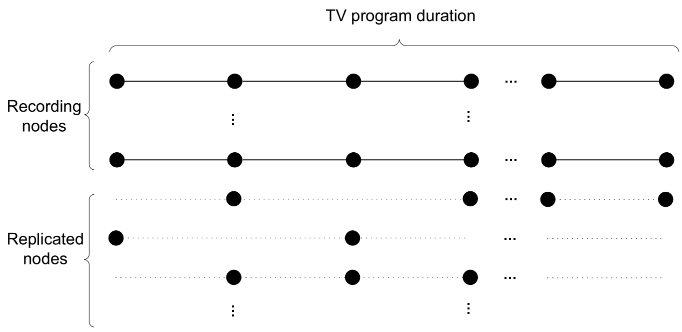
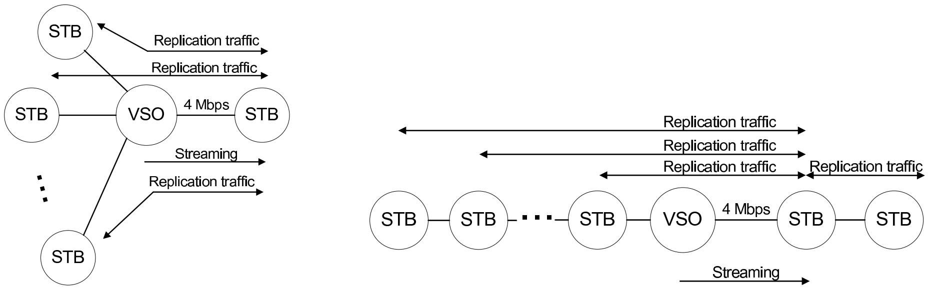



Figure 5.22: Simulation scenario for MDRS

Comparative simulation tests are performed as described in the previous section in order to evaluate the performance of the proposed MRDS. There are two main aspects to be analysed: data and network. In terms of data, the performance reflects whether the required data is accessible and what the overhead is for maintaining data accessibility. In terms of network resource usage for data replication, data loss during delivery is used to measure the performance. Additionally, streaming tests address which combination of data replication and streaming approach shows better performance.

Table 5.12 and 5.13 present the results of the comparison between the Multimedia Data Replication Scheme (MDRS) and the other two approaches: AR and Tempo. The assessment is performed in terms of the following metrics: number of data segments (Num.), average data set size (D.S.) and average data availability (D.A.). As assumed and stated in Table 5.7, the device availability is 0.4. With this value, the minimum data set can be simply




(a) Star topology (b) String topology

Figure 5.23: Network topology used for streaming simulation (4 Mbps downlink, 220 kbps uplink)

| Recording Node(s) | MRDS | | | AR | | | Tempo | | |
|---|---|---|---|---|---|---|---|---|---|
| | Num. | D.S. | D.A. | Num. | D.S. | D.A. | Num. | D.S. | D.A. |
| 1 | 192 | 1.9 | 0.729 | 595 | 5.8 | 0.626 | 842 | 8.25 | 0.428 |
| 2 | 238 | 2.3 | 0.747 | 686 | 6.7 | 0.654 | 1098 | 10.8 | 0.476 |
| 3 | 306 | 3 | 0.980 | 604 | 5.9 | 0.726 | 1386 | 13.6 | 0.541 |
| 4 | 408 | 4 | 0.980 | 408 | 4 | 0.980 | 1543 | 15.1 | 0.557 |
| 5 | 510 | 5 | 0.980 | 510 | 5 | 0.980 | 1631 | 16.0 | 0.592 |
| 6 | 612 | 6 | 0.980 | 612 | 6 | 0.980 | 1771 | 17.4 | 0.629 |
| 7 | 714 | 7 | 0.980 | 714 | 7 | 0.980 | 1856 | 18.2 | 0.662 |
| 10 | 1020 | 10 | 0.980 | 1020 | 10 | 0.980 | 2023 | 19.8 | 0.739 |
| 15 | 1530 | 15 | 0.980 | 1530 | 15 | 0.980 | 2040 | 20 | 0.934 |

Table 5.12: Simulation results statistics in star topology (simulation time 2 hours)

| Recording node(s) | MRDS | | | AR | | | Tempo | | |
|---|---|---|---|---|---|---|---|---|---|
| | Num. | D.S. | D.A. | Num. | D.S. | D.A. | Num. | D.S. | D.A. |
| 1 | 283 | 2.8 | 0.882 | 1059 | 10.4 | 0.576 | 911 | 8.9 | 0.510 |
| 2 | 331 | 3.2 | 0.943 | 1051 | 10.3 | 0.733 | 1116 | 10.9 | 0.538 |
| 3 | 306 | 3 | 0.980 | 772 | 7.6 | 0.900 | 1393 | 13.7 | 0.596 |
| 4 | 408 | 4 | 0.980 | 408 | 4 | 0.980 | 1496 | 14.7 | 0.608 |
| 5 | 510 | 5 | 0.980 | 510 | 5 | 0.980 | 1707 | 16.7 | 0.652 |
| 6 | 612 | 6 | 0.980 | 612 | 6 | 0.980 | 1494 | 14.6 | 0.682 |
| 7 | 714 | 7 | 0.980 | 714 | 7 | 0.980 | 1500 | 14.7 | 0.729 |
| 10 | 1020 | 10 | 0.980 | 1020 | 10 | 0.980 | 1806 | 17.7 | 0.780 |
| 15 | 1530 | 15 | 0.980 | 1530 | 15 | 0.980 | 1975 | 19.4 | 0.900 |

Table 5.13: Simulation results statistics in string topology (simulation time 2 hours)

calculated by Eq. 4.14. In general, the number of data (Num.) and average data set size (D.S.) expand as the number of recording applications increases. The basic reason for this tendency is simply that the initial number of data replications could be considered as the number of segments multiplied by the number of recording applications. MDRS and AR show that there is no data replication when the number of recording applications is bigger than 3 for AR and 2 for MDRS. However, Tempo does not show saturation for the number of data and data sets during the simulation since that considers only the target network usage and any numbers related to data are not used in the algorithm for replication.

In relation to the star topology, the simulation results present interesting values of MRDS when the recording application number is 1 and 2. Since recording applications provide a lower number of segments than the minimum data set (e.g. 3), the nodes which contain the data try to duplicate it into different devices. AR also shows similar results in this topology. However, as it is based on estimated data availability, it tries to replicate more data into other devices than required, unnecessarily loading the network and consuming additional storage space. As generally discussed, Tempo shows no saturation and continuously replicates data during the simulation.

Test results, for the string topology, show similar characteristics to those for the star topology. MRDS and AR perform no data replication with 2 and 3 recording nodes, respectively. As the number of data in MRDS is targeting the minimum data set (e.g. 3), the replication testing results are very close to this value. On other hand, AR shows higher data replication than MRDS because its estimated data availability is quite low and more data is needed in order to achieve a higher value. Tempo shows results similar to those obtained for the star topology.

Fig. 5.24 and 5.25 show the results assessed by network usage and data loss in each topology. There is no drop of packets during the simulation of the string topology. The assumed string topology includes asynchronous connections among the nodes. Downstream, which has a higher bandwidth compared to the connection in the opposite direction, is mainly used for delivering replication data in the current simulation environment and data sources are mostly located in the nodes utilising the downstream connections. Therefore, data loss of string topology is not presented. Generally, network usage increases with the number of recording applications. However, MRDS and AR each have saturation points since the amount of replicated data is determined based on their own restrictions, as monitored in Fig. 5.25(a) and 5.25(b).

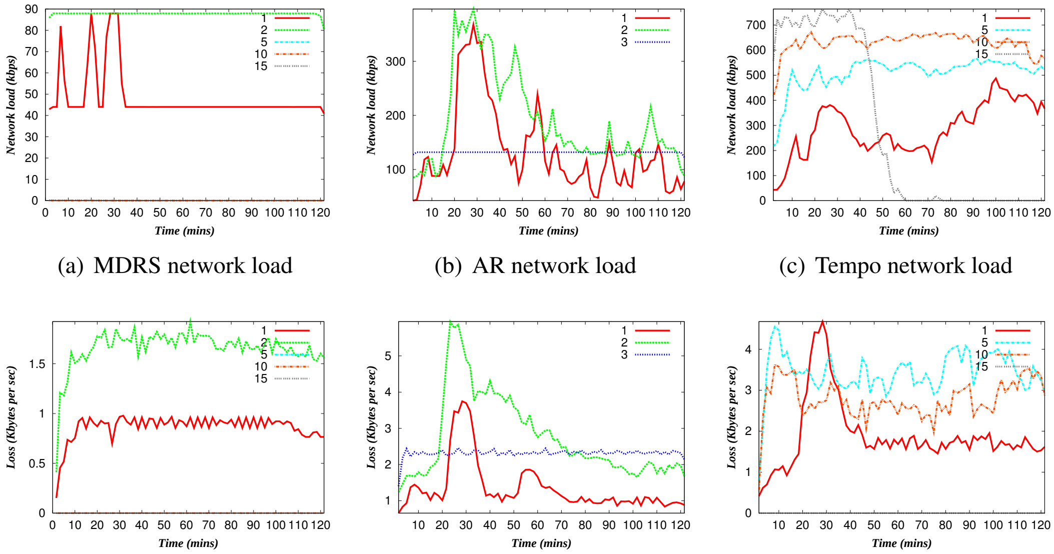
Fig. 5.24(a) shows MDRS network usage. Network usage increases with the number of recording applications. The minimum data set 3 is achieved after the recording application number equals 3. Due to this, other cases with larger numbers of recording applications show no data replication. Network usage of AR in the string topology is similar to this situation, increasing until 3 recording applications are reached. After that, there is no data replication. In contrast, Tempo shows a continuous increase with the number of recording applications. In the string topology, network usage presents similar tendencies for each scheme.

Fig. 5.24(a), 5.24(b) and 5.24(c) illustrate network load when the star topology was employed. MDRS uses 109 kbps on average when 2 recording nodes are involved, whereas AR and Tempo use 239 kbps and 536 kbps, respectively. These numbers are more than twice that of MDRS. In terms of data loss, MDRS shows on average 1.7 Kbytes per sec with 2 recording nodes. Meanwhile, AR presents 2.8 Kbytes per sec and Tempo shows 2.4 Kbytes per sec on average. These results show much better performance in terms of network load and loss rate for the proposed MDRS in comparison with AR and Tempo.

Table 5.14 and Table 5.15 present throughput and estimated Peak Signal-to-Noise Ratio (PSNR) during TFRC and UDP-based multimedia streaming, comparing the proposed scheme MRDS with two other solutions: AR and Tempo. Estimated PSNR is measured based on the throughput and loss during streaming, and is expressed in decibels.

In general, MRDS provides a higher throughput and estimated PSNR in comparison with the other data replication approaches tested. UDP based streaming shows high and relatively stable throughput since it does not involve any adaptation. TFRC provides adaptation using rate control. In order to do this, the receiver generates feedback and sends it to the sender. As the 220 kbps upstream channel is easily filled with data replication traffic, TFRC tries to reduce the multimedia transmission rate affecting user perceived quality.

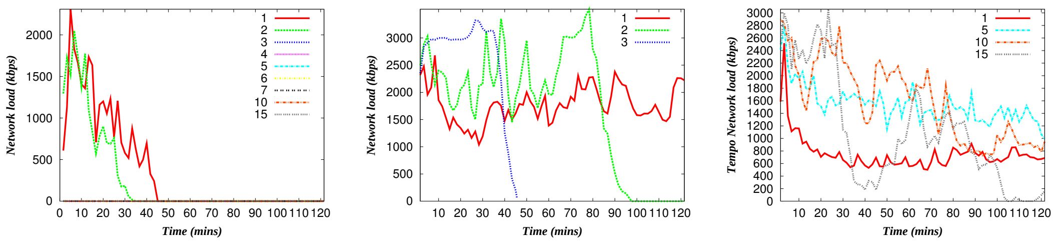
Table 5.14 and Table 5.15 include testing results when star and string topologies are




(a) MDRS network load (b) AR network load (c) Tempo network load

(d) MDRS data loss (e) AR data loss (f) Tempo data loss

Figure 5.24: Network load and data loss in star topology




(a) MDRS network load (b) AR network load (c) Tempo network load

Figure 5.25: Network load in string topology

used, respectively. In addition, Fig. 5.26 and Fig. 5.27 presents bar chart of estimated PSNR when star and string topologies are used, respectively. When using UDP streaming over the star topology, MDRS shows 8.78% and 18.88% improvement against AR and Tempo in terms of estimated PSNR. When employing UDP over the string topology, similar PSNR is obtained in all three schemes. A significant benefit of using MDRS is when transmitting using TFRC. In this situation, PSNR is 8 and 4 times higher than AR and Tempo, respectively. When TFRC streaming is applied over the string topology, MDRS shows 30.65% and 71.14% improvement against AR and Tempo in terms of estimated PSNR, respectively.

In summary, the proposed MRDS is compared against other general purpose data repli-

| Underlying protocol | Replication scheme | Streaming throughput (Mbps) | PSNR (dB) |
|---|---|---|---|
| UDP | MDRS | 3.98 | 46.98 |
| | AR | 3.97 | 43.19 |
| | Tempo | 3.96 | 39.53 |
| TFRC | MDRS | 3.94 | 36.86 |
| | AR | 1.43 | 3.86 |
| | Tempo | 2.25 | 7.16 |

Table 5.14: Results for multimedia streaming over the Star topology

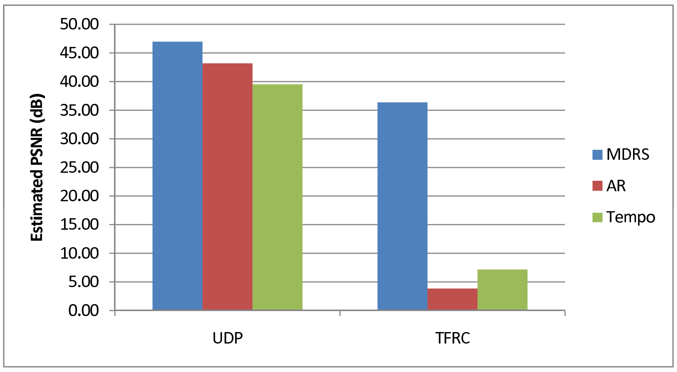



Figure 5.26: Estimated PSNR of multimedia streaming over the Star topology

| Underlying protocol | Replication scheme | Streaming throughput (Mbps) | PSNR (dB) |
|---|---|---|---|
| UDP | MDRS | 3.98 | 45.90 |
| | AR | 3.98 | 44.80 |
| | Tempo | 3.98 | 45.90 |
| TFRC | MDRS | 3.93 | 34.57 |
| | AR | 3.81 | 26.46 |
| | Tempo | 3.61 | 20.20 |

Table 5.15: Results for multimedia streaming over the String topology

cation schemes such as AR and Tempo. The tests include scenarios with different numbers of recording nodes. MRDS shows better performance than the other general purpose data replication schemes in terms of data availability and network resource usage on different typical topologies for broadband service networks. When streaming is performed with MRDS data replication, better performance in terms of throughput and estimated PSNR in

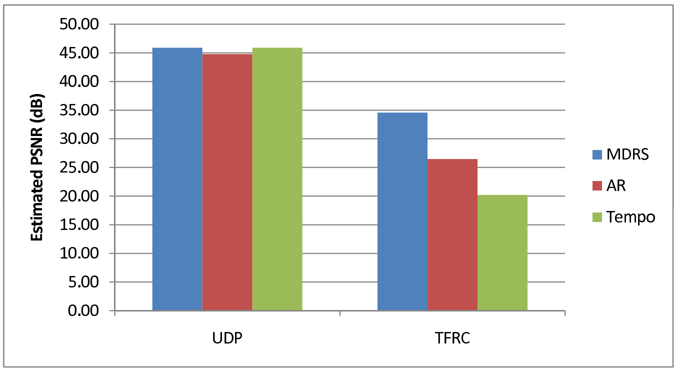



Figure 5.27: Estimated PSNR of multimedia streaming over the String topology

comparison with other two approaches is achieved.

### 5.4.3 Multiple-Source Streaming Test Results and Analysis

#### 5.4.3.1 Assumptions for Simulation Tests

The test scenarios for multiple-source streaming use the clips mentioned in section 5.2.4.1 which are selected at random by each receiver. The duration of the simulation is set to 100 secs. All connections start at 1 sec after the simulation start. The scenario includes 5 FTP connections as background traffic which start and end like the streaming connections. The QAMMD, mTFRC and mUDP approaches are used in turn as multiple source streaming methods. The number of clients is gradually increased in steps of 10. Since here a receiver is assumed to have three senders, the total number of senders is three times as large as the number of receivers. For example, 10 receivers require 30 nodes as senders. Consequently, 40 nodes are involved in the test for 10 receivers.

As discussed in section 5.4.1, the main performance assessment metric for streaming simulation in this thesis is PSNR. The following sections present testing result when each of the topologies introduced are used in turn: wired, wireless and WLAN-WiMAX dumbbell topologies.

#### 5.4.3.2 Wired Dumbbell Topology

In uncongested conditions (between 10 and 60 nodes in the system), all approaches show no packet loss. However, loss with mUDP is significantly increased when congestion builds up. Relatively, TFRC-based approaches including QAMMD and mTFRC experience less loss, with QAMMD outperforming the other solutions at all times.

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 10 | 39.3 | 38.6 | 52.6 | 61.1 |
| 20 | 42.2 | 42.8 | 54.7 | 74.9 |
| 30 | 40.5 | 41.5 | 50.2 | 81.4 |
| 40 | 40.9 | 41.3 | 51.9 | 82.3 |
| 50 | 39.8 | 39.0 | 49.8 | 80.7 |
| 60 | 39.4 | 38.0 | 53.1 | 73.7 |
| Avg. | 40.4 | 40.2 | 52.1 | 75.7 |

Table 5.16: Estimated PSNR with wired dumbbell topology (dB)

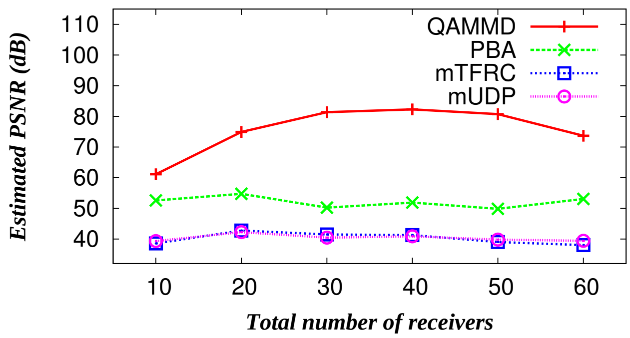



Figure 5.28: Comparison between estimated PSNR with various solutions

Table 5.16 and Fig. 5.28 show a comparison between schemes in terms of quality, as estimated by Peak Signal Noise Ratio (PSNR) using Eq. 5.3 based on frame loss and throughput with increasing numbers of users. On average, when using QAMMD, PSNR is 75.7 dB, whereas when PBA, mTFRC and mUDP are employed PSNR is 52.1 dB, 40.2 dB and 40.4 dB, respectively. It can be seen how QAMMD behaves with 45.3% better than PBA, with 88.3% better than mTFRC and with 87.4% better than mUDP. Since these results are for uncongested conditions, PSNR values fluctuate against the number of nodes. Specifically, when the bottleneck channel is crowded with 60 nodes, QAMMD offers 38.8%

better perceived quality than PBA, 93.9% better perceived quality than mTFRC and 87.1% better than mUDP expressed in terms of PSNR.

As another performance metric, buffer underflow is compared in Table 5.17 and Fig. 5.29 between the schemes. On average, when using QAMMD, buffer underflow is 0.7, whereas when PBA, mTFRC and mUDP are employed buffer underflow is 1.2, 4.3 and 5.1, respectively. It can be seen how QAMMD has 41.7% less buffer underflow than PBA. In addition, mTFRC and mUDP show almost 6.1 and 7.3 times more buffer underflow events than QAMMD, respectively.

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 10 | 5.60 | 5.00 | 1.20 | 0.10 |
| 20 | 4.15 | 3.45 | 1.10 | 0.05 |
| 30 | 4.77 | 4.07 | 1.40 | 0.03 |
| 40 | 4.85 | 4.08 | 1.25 | 0.03 |
| 50 | 5.36 | 4.54 | 1.36 | 0.04 |
| 60 | 5.60 | 4.90 | 0.92 | 0.05 |
| Avg. | 5.05 | 4.34 | 1.20 | 0.05 |

Table 5.17: Buffer underflow with wired dumbbell topology (times/node)

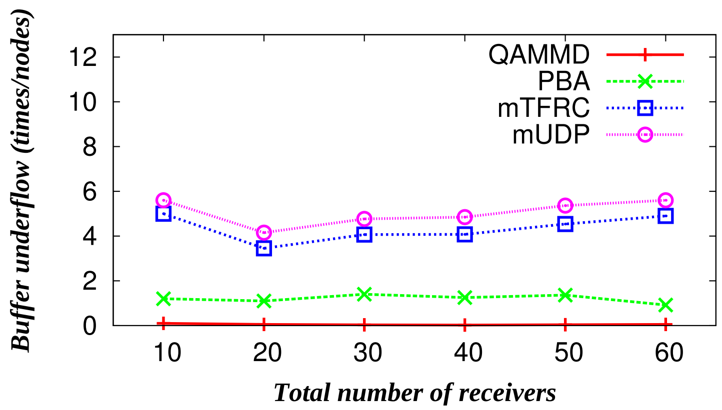



Figure 5.29: Comparison between buffer underflow with various solutions

Average initial waiting time and average total waiting time as overheads are presented in Table 5.18, Fig 5.30, Table 5.19 and Fig 5.31, respectively. On average, when using QAMMD, initial waiting time is 16.5 secs, whereas when PBA, mTFRC and mUDP are employed the initial waiting time is 1.6 secs, 0.6 secs and 0.6 secs, respectively. However, the total waiting time dramatically changes. When using QAMMD, total waiting time is 19.7 secs, whereas when PBA, mTFRC and mUDP are employed total waiting time is 43.2 secs, 169.0 secs and 202.4 secs, respectively.

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 10 | 0.6 | 0.6 | 1.2 | 16.1 |
| 20 | 0.6 | 0.6 | 1.1 | 16.3 |
| 30 | 0.6 | 0.6 | 0.9 | 16.6 |
| 40 | 0.6 | 0.6 | 1.1 | 16.6 |
| 50 | 0.6 | 0.6 | 1.3 | 16.2 |
| 60 | 0.6 | 0.6 | 4.1 | 17.2 |
| Avg. | 0.6 | 0.6 | 1.6 | 16.5 |

Table 5.18: Initial waiting time with wired dumbbell topology (secs)

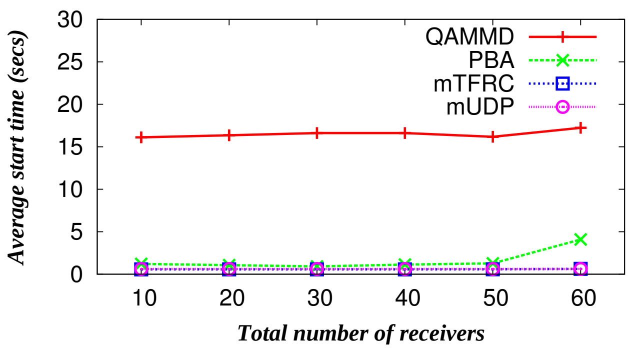



Figure 5.30: Comparison between average initial waiting time with various solutions

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 10 | 160.9 | 138.7 | 35.0 | 23.1 |
| 20 | 195.9 | 164.0 | 34.9 | 19.8 |
| 30 | 181.0 | 147.2 | 51.3 | 18.9 |
| 40 | 183.7 | 148.5 | 44.9 | 18.3 |
| 50 | 231.8 | 190.2 | 50.7 | 18.6 |
| 60 | 261.4 | 225.4 | 42.2 | 19.6 |
| Avg. | 202.4 | 169.0 | 43.2 | 19.7 |

Table 5.19: Total waiting time with wired dumbbell topology (secs)

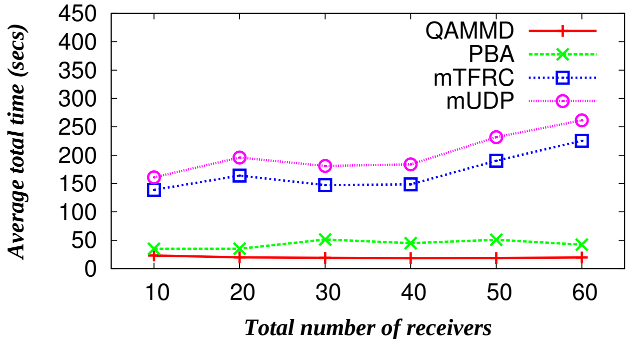



Figure 5.31: Comparison between average total waiting time with various solutions

#### 5.4.3.3 Wireless Dumbbell Topology

The wireless dumbbell topology as depicted in Fig. 5.11(b), is used for the tests in this section. The number of receiving nodes is limited up to 4 since 17.4 Mbps and 13.2 Mbps are the maximum achievable throughputs using UDP and TCP over the wireless dumbbell topology. The test includes simulation results when 10 different start points are selected from each movie in order to achieve stronger statistical results. PSNR, buffer underflow and average waiting time for the wireless dumbbell topology will be discussed in this section.

Table 5.20 and Fig. 5.32 show a comparison between schemes in terms of estimated PSNR with increasing numbers of users. On average, when using QAMMD, PSNR is 60.9 dB, whereas when PBA, mTFRC and mUDP are employed PSNR is 51.0 dB, 38.1 dB and 44.8 dB, respectively. It can be seen how QAMMD behaves with 19.4% better than PBA, with 59.8% better than mTFRC and with 35.9% better than mUDP. Specifically, when the wireless bottleneck channel is crowded with 4 nodes, QAMMD offers 42.3% better perceived quality than PBA, 90.7% better perceived quality than mTFRC and 79.5% better than mUDP expressed in terms of PSNR.

Buffer underflow is compared in Table 5.21 and Fig. 5.33 between the schemes. On average, when using QAMMD, buffer underflow is 0.07, whereas when PBA, mTFRC and mUDP are employed buffer underflow is 1.17, 5.86 and 4.52, respectively. It can be seen how QAMMD has 94.0 % less buffer underflow events than PBA. In addition, mTFRC and mUDP show almost 83.7 and 65.6 times more buffer underflow events than QAMMD,

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 1 | 65.1 | 47.4 | 62.8 | 62.5 |
| 2 | 37.8 | 32.1 | 54.7 | 58.5 |
| 3 | 42.2 | 40.8 | 43.5 | 61.3 |
| 4 | 34.1 | 32.1 | 43.0 | 61.2 |
| Avg. | 44.8 | 38.1 | 51.0 | 60.9 |

Table 5.20: Estimated PSNR with wireless dumbbell topology (dB)

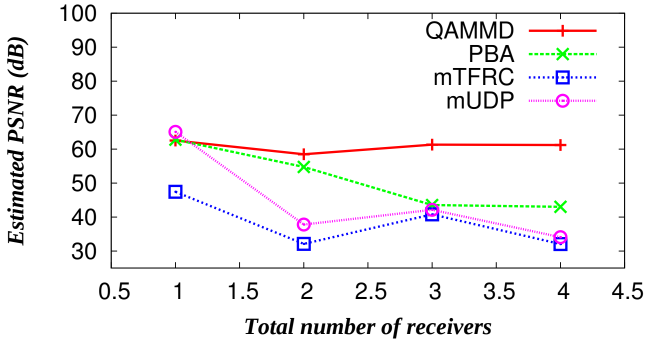



Figure 5.32: Comparison between Estimated PSNR with Various Solutions

respectively.

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 1 | 0.00 | 0.92 | 0.00 | 0.00 |
| 2 | 5.21 | 11.83 | 0.71 | 0.25 |
| 3 | 3.75 | 2.67 | 2.06 | 0.00 |
| 4 | 9.10 | 8.02 | 1.92 | 0.04 |
| Avg. | 4.52 | 5.86 | 1.17 | 0.07 |

Table 5.21: Buffer underflow with wireless dumbbell topology (times/node)

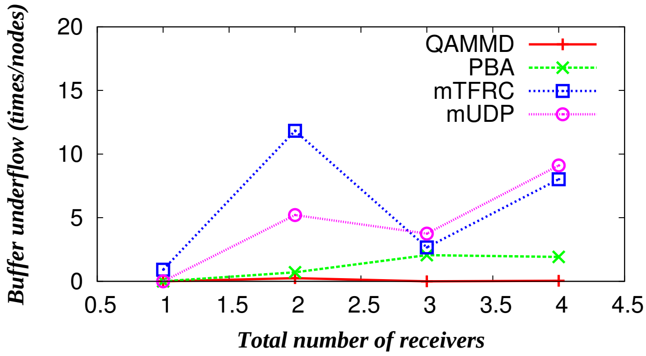



Figure 5.33: Comparison between buffer underflow with Various Solutions

Average initial waiting time and average total waiting time as overheads are presented in Fig 5.34 and Fig 5.35, respectively. On average, when using QAMMD, initial waiting time is 19.3 secs, whereas when PBA, mTFRC and mUDP are employed initial waiting time is 15.3 secs, 1.6 secs and 0.6 secs, respectively. However, total waiting time dramatically changes. When using QAMMD, total waiting time is 23.8 secs, whereas when PBA, mTFRC and mUDP are employed total waiting time is 64.3 secs, 238.3 secs and 213.8 secs, respectively.

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 1 | 0.6 | 1.7 | 19.8 | 19.8 |
| 2 | 0.6 | 1.7 | 19.7 | 19.7 |
| 3 | 0.6 | 1.4 | 18.6 | 18.6 |
| 4 | 0.6 | 1.4 | 19.0 | 19.0 |
| Avg. | 0.6 | 1.6 | 19.3 | 19.3 |

Table 5.22: Initial waiting time with wireless dumbbell topology (secs)

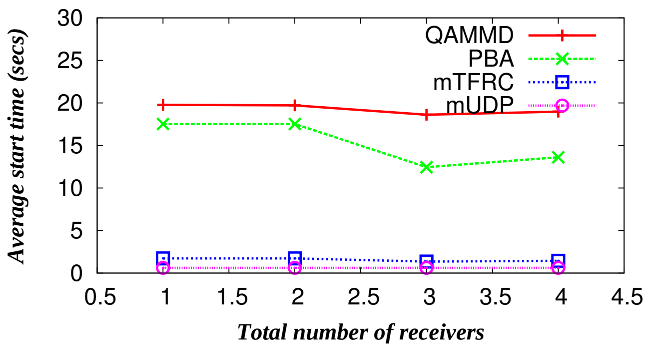



Figure 5.34: Comparison between average initial waiting time with various solutions

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 1 | 0.6 | 5.3 | 17.5 | 19.8 |
| 2 | 281.3 | 557.6 | 52.5 | 33.6 |
| 3 | 168.2 | 64.9 | 78.1 | 18.6 |
| 4 | 405.0 | 325.6 | 109.0 | 23.3 |
| Avg. | 213.8 | 238.3 | 64.3 | 23.8 |

Table 5.23: Total waiting time with wireless dumbbell topology (secs)

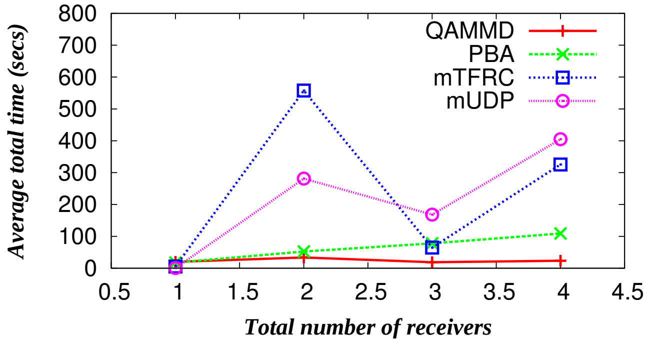



Figure 5.35: Comparison between average total waiting time with various solutions

#### 5.4.3.4 WLAN-WiMAX Dumbbell Topology

The WLAN-WiMAX dumbbell topology (See Fig. 5.11(c)) is used for the tests in this section. There are only up to 3 receiving nodes available because of lower bandwidth from WiMAX. The test includes simulation results of 10 different start points of each movie similar to wireless topology in order to have realistic statistics. In addition, the test scenario includes a simple mobility scenario which is presented in Fig. 5.36. Receivers start streaming within the coverage of WiMAX, move into the coverage of WLAN and go out to the WiMAX coverage again. The start time of the receiver movement varies from 12 secs to 22 secs in a uniform distribution. Similar to the wired dumbbell topology, PSNR, buffer underflow and average waiting time will be discussed in this section.

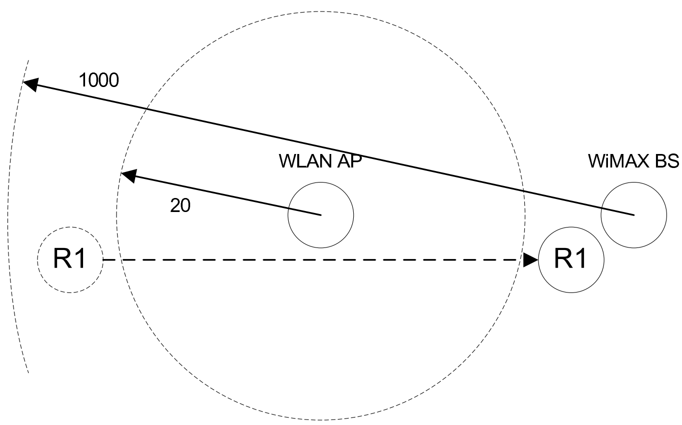



Figure 5.36: WLAN-WiMAX dumbbell scenario based on MIH

Table 5.24 and Fig. 5.37 show a comparison between schemes in terms of estimated PSNR with increasing numbers of users. On average, when using QAMMD, PSNR is 84.2 dB, whereas when PBA, mTFRC and mUDP are employed PSNR is 70.8 dB, 56.2 dB and 57.0 dB, respectively. It can be seen how QAMMD behaves with 18.9% better than PBA, with 49.8% better than mTFRC and with 47.7% better than mUDP. Specifically, when the wireless bottleneck channel is crowded with 3 nodes, QAMMD offers 68.5% better perceived quality than PBA, 112.3% better perceived quality than mTFRC and 111.3% better than mUDP, expressed in terms of PSNR.

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 1 | 90.0 | 90.0 | 90.0 | 90.0 |
| 2 | 38.3 | 36.0 | 68.9 | 72.5 |
| 3 | 42.6 | 42.4 | 53.4 | 90.0 |
| Avg. | 57.0 | 56.2 | 70.8 | 84.2 |

Table 5.24: Estimated PSNR with WLAN-WiMAX dumbbell topology (dB)

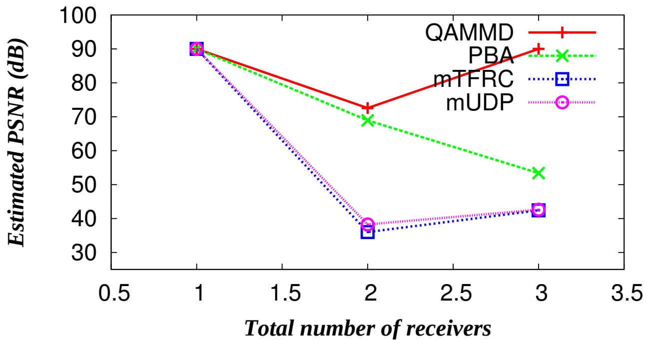



Figure 5.37: Comparison between Estimated PSNR with Various Solutions

Buffer underflow is compared in Table 5.25 and Fig. 5.38 between the schemes. On average, when using QAMMD, buffer underflow events are 0.03, whereas when PBA, mTFRC and mUDP are employed buffer underflow metric is 0.23, 2.57 and 2.99, respectively. It can be seen how QAMMD behaves with 87.0 % better than PBA. In addition, mTFRC and mUDP show almost 85.7 and 99.7 times more buffer underflow events than QAMMD, respectively.

Average initial waiting time and average total waiting time as overheads are presented

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 1 | 0.00 | 0.00 | 0.00 | 0.00 |
| 2 | 5.21 | 5.13 | 0.13 | 0.08 |
| 3 | 3.75 | 2.58 | 0.56 | 0.00 |
| Avg. | 2.99 | 2.57 | 0.23 | 0.03 |

Table 5.25: Buffer underflow with WLAN-WiMAX dumbbell topology (times/node)

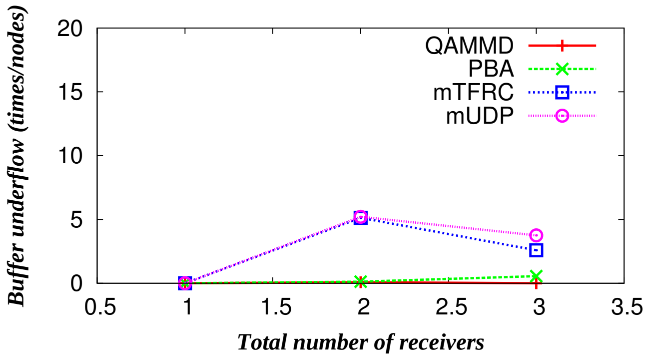



Figure 5.38: Comparison between buffer underflow with Various Solutions

in Table 5.26, Fig 5.39, Table 5.27 and Fig 5.40, respectively. On average, when using QAMMD, initial waiting time is 17.3 secs, whereas when PBA, mTFRC and mUDP are employed initial waiting time is 6.9 secs, 0.7 secs and 0.6 secs, respectively. However, total waiting time dramatically changes again. When using QAMMD, total waiting time is 18.4 secs, whereas when mTFRC and mUDP are employed total waiting time is 123.4 secs and 150.0 secs, respectively. Only the case of PBA is shorter than QAMMD with an average of total waiting time of 17.0 secs. However, QAMMD shows shorter total waiting time for two and three nodes.

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 1 | 0.6 | 0.7 | 4.1 | 16.7 |
| 2 | 0.6 | 0.7 | 9.2 | 17.6 |
| 3 | 0.6 | 0.7 | 8.4 | 17.6 |
| Avg. | 0.6 | 0.7 | 7.2 | 17.3 |

Table 5.26: Initial waiting time with WLAN-WiMAX dumbbell topology (secs)

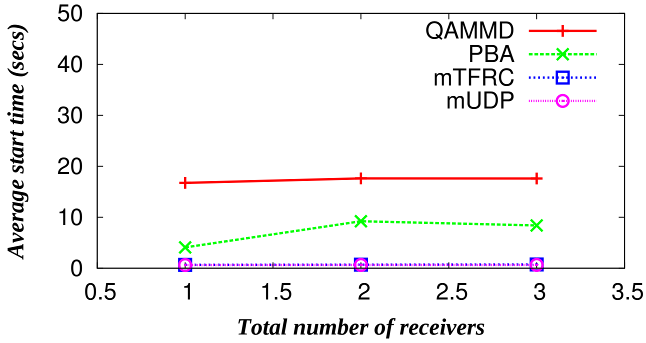



Figure 5.39: Comparison between average initial waiting time with various solutions

| Num. of nodes | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|
| 1 | 0.6 | 0.7 | 4.1 | 16.7 |
| 2 | 280.9 | 277.6 | 17.3 | 20.7 |
| 3 | 168.4 | 92.0 | 29.5 | 17.6 |
| Avg. | 150.0 | 123.4 | 17.0 | 18.4 |

Table 5.27: Total waiting time with WLAN-WiMAX dumbbell topology (secs)

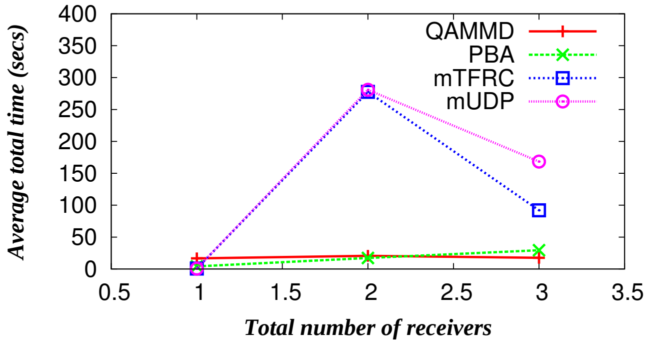



Figure 5.40: Comparison between average total waiting time with various solutions

## 5.5 Chapter Summary

This chapter presents modelling of Smart PIN focus on its two novel solutions: Multimedia Data Replication Scheme (MDRS) and Quality Adaptive Multiple-source Multimedia Delivery (QAMMD). The model includes not only the service component and applications but also Media Independent Handover (MIH) as a virtually always connected heterogeneous network model. All models are based on Network Simulator 2.31.

MDRS is compared with Autonomous Replication (AR) and Tempo, which are autonomous data replication schemes based on data availability and network resource usage control. MDRS shows better performance in terms of data availability and network resource usage especially with information overload and segmented data sharing.

In terms of multiple-source streaming, QAMMD is compared with a multiple TFRC connection-based scheme (mTFRC) and a multiple UDP connection based scheme (mUDP). In addition, a similar buffer estimation algorithm, the Predictive Buffer Algorithm (PBA) is used in simulations. The performance is evaluated for multiple-source streaming approaches in terms of PSNR and overhead measurements. QAMMD shows better performance in terms of estimated PSNR, buffer underflow and total waiting using different network topologies.

In the next chapter, the prototyping of Smart PIN will be discussed. It focuses on multiple-source streaming while the quality of multimedia will be assessed in detail.

# Chapter 6

# Prototyping and Results

## 6.1 Introduction

Smart PIN includes the proposed novel data replication and multimedia streaming features. The previous simulations for Smart PIN considered network performance related to issues of data replication and multimedia streaming with network usage, throughput, loss, etc. Although the Peak Signal to Noise Ratio (PSNR) was presented as an estimation of user quality measurement, the actual user perceived quality of delivered multimedia requires a different setup from simulations. Prototyping Smart PIN as reported in this chapter and the related real life-like test will assess system performance and user perceived quality of delivered multimedia, when proposed solutions are deployed.

An evaluation of network performance with data replication requires very long duration as well as many resources such as PCs and network components. Instead of implementing data replication, a background traffic based on TCP protocol is used to model data replication traffic in the prototyping-based tests reported in this chapter. In addition, the UDP protocol is used since similar multimedia-based network traffic can be found in real networks.

In order to measure user perceived quality, multiple-source streaming based on the proposed Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD) algorithm is implemented in order to have packet scheduling among the multiple senders. The current testbed uses two servers and one client in order to focus on buffer estimation and management as illustrated in Fig. 6.1. Detailed algorithms for buffer estimation and buffer management can be found in 4.4. These tests uses different transport protocols including UDP and TFRC.

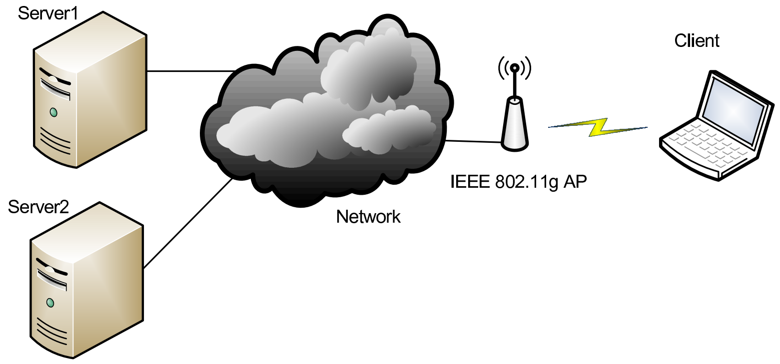



Figure 6.1: Smart PIN prototyping concept

As mentioned earlier, prototyping requires a TFRC protocol and a multimedia streaming application. For implementation efficiency, existing solutions are adopted as much as possible, although there are only few available implementations and most of these are at an experimental stage. One of those implementations is **DCCP CCID 3** in a Linux kernel which is easy to use through a socket programming interface[1]. In addition, there is a streaming application, **Video LAN**[2] that supports DCCP on Linux.

This chapter is structured as follows. Firstly, the prototyping architecture and implementation details related to the Video LAN prototyping system is presented, supporting buffer estimation and management. In order to measure network performance and the user perceived quality of delivered multimedia, the test systems we use for networking tests and user tests are discussed next. Finally, test results and analysis of the networking tests and user perceptual tests are discussed separately. This is followed by the chapter conclusion.

[1]Net:DCCP - The Linux Foundation: http://www.linuxfoundation.org/en/Net:DCCP, last accessed 18 Nov. 2009

[2]VideoLAN - VLC media player, http://www.videolan.org/, last accessed 18 Nov. 2009

## 6.2 Prototyping Architecture and Implementation Details

### 6.2.1 Introduction to VLC

The Video LAN Client (VLC) is an open-source multimedia player, freely available from the internet. It has been ported onto various platforms such as Microsoft Windows, Linux, etc., supporting various multimedia formats and streaming. The Smart PIN prototyping system uses VLC for servers and a client running on Linux. Currently, the VLC 0.9.8a version is used on a Debian variant Linux, Ubuntu [3] 8.10 (Linux kernel 2.6.27-11) for overall testing on Pentium 4 processor computers.

On the Linux platform, VLC supports DCCP as a transport protocol which includes the TFRC option (i.e. DCCP CCID-3) for congestion control. At the application level, VLC uses RTP in order to deliver multimedia data when DCCP is used. RTP could be delivered over UDP but the test procedures are different from DCCP.

Since VLC is multimedia player, it supports local file play. In addition, VLC supports streaming, too. In order to provide understanding of VLC operation, following subsections describes module chains for local file play, single server streaming. After that, modification of VLC for multiple-source streaming follows.

### 6.2.2 VLC Operation for Local File Play

The general architecture of VLC is depicted in Fig 6.2 for playing a MPEG2 file which has MPEG2 Program Stream (MPEG2 PS) format. VLC includes a group of modules such as interface, input, video decoder, audio decoder, video output and audio output. In each group, the main module initiates the required module on-the-fly in order to process the required multimedia format such as MPEG2, MPEG4, etc. During the playing of an MPEG2 movie, an *MPEG2 PS demux* is initiated and is involved in separating the video and audio streams. Each stream is then handed to the proper decoder modules, namely *video decoder* and *audio decoder*.

[3] Ubunutu, http://www.ubuntu.com/, last accessed 18 Nov. 2009

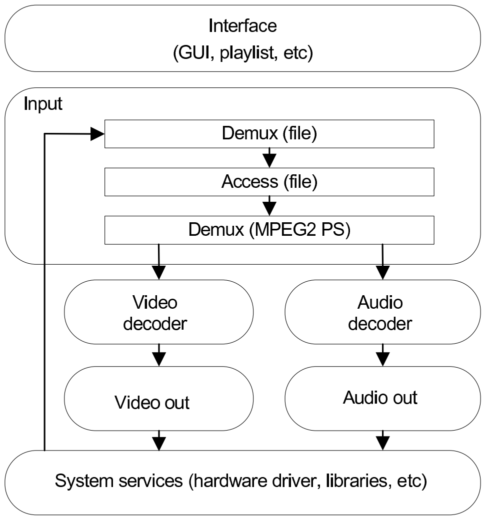



Figure 6.2: VLC module chain for MPEG2 movie file playback

VLC modules basically share a single thread. In addition, some of the features in VLC are implemented as threads, optimising the overall executions. In such cases, conceptually the threads run in parallel. In order to support communication between threads, a First-In First-Out (FIFO) queue is used in VLC.

### 6.2.3 VLC Operation for Single Server Streaming

VLC modules for streaming are presented in Fig. 6.3. When it is used for streaming, the MPEG2 file (MPEG2 PS format) is converted into MPEG2 Transport Stream (MPEG2 TS) format and is delivered over RTP to the client. When data reaches the client, the *RTP demux* extracts the MPEG stream and passes it to the *Stream* module.

The *Stream* module is implemented as a thread and communicates with the *RTP demux* through a FIFO queue. In the *Stream* thread, the *MPEG2 TS demux* module is used for de-multiplexing of elementary streams such as MPEG video and audio. The proposed **Buffer Underflow Avoidance Scheme (BUAS)** is included in *RTP demux* and *Stream* thread since they have a queue between them.

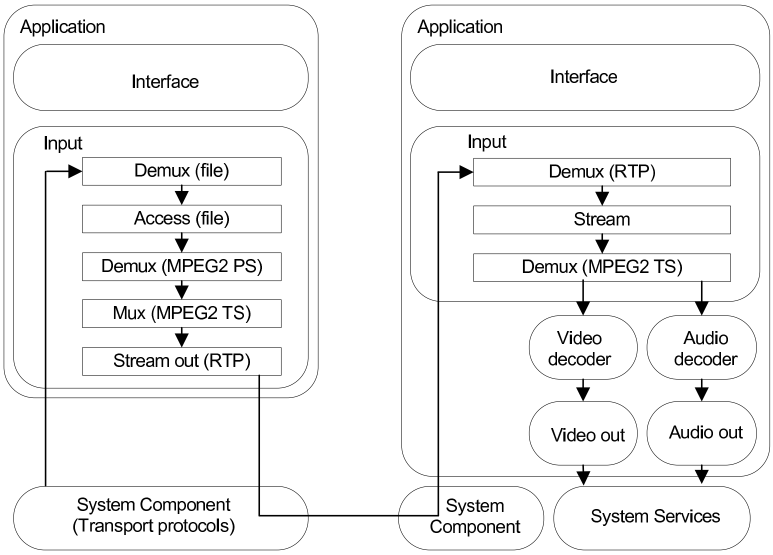



Figure 6.3: VLC module chain for MPEG2 movie streaming

### 6.2.4 VLC Modification for Multiple-source Streaming

Since VLC does not support multiple-source streaming, structure modifications include connection establishment, packet scheduling, and sender synchronisation. As a part of contributions of this thesis, these modifications are applied mainly in *Mux* in senders, *Demux* in the receiver, and *Stream* module in senders and a receiver as depicted in Fig 6.4. Although they mention that VLC supports MPEG2 PS streaming, the receiver implementation does not detect MPEG2 PS streaming through the network. In order to support this, the Stream module in the receiver also needs to be modified. In addition, a bandwidth limitation is also applied on the server side in order to achieve similar conditions for our simulation.

The modifications for **connection establishment** include multiple *Mux* creation based on the number of servers. In addition, receiver-based connection setup is used for multiple-source streaming. VLC establishes the connection between a sender and a receiver in different ways depending on the transport protocol. If a connection-based protocol such as TCP or DCCP is used, the receiver initiates a connection and starts the sender's data transmission. If a connection-less protocol (i.e. UDP) is used, the sender transmits data before the receiver starts to receive data. In case of multiple-source streaming, the receiver has

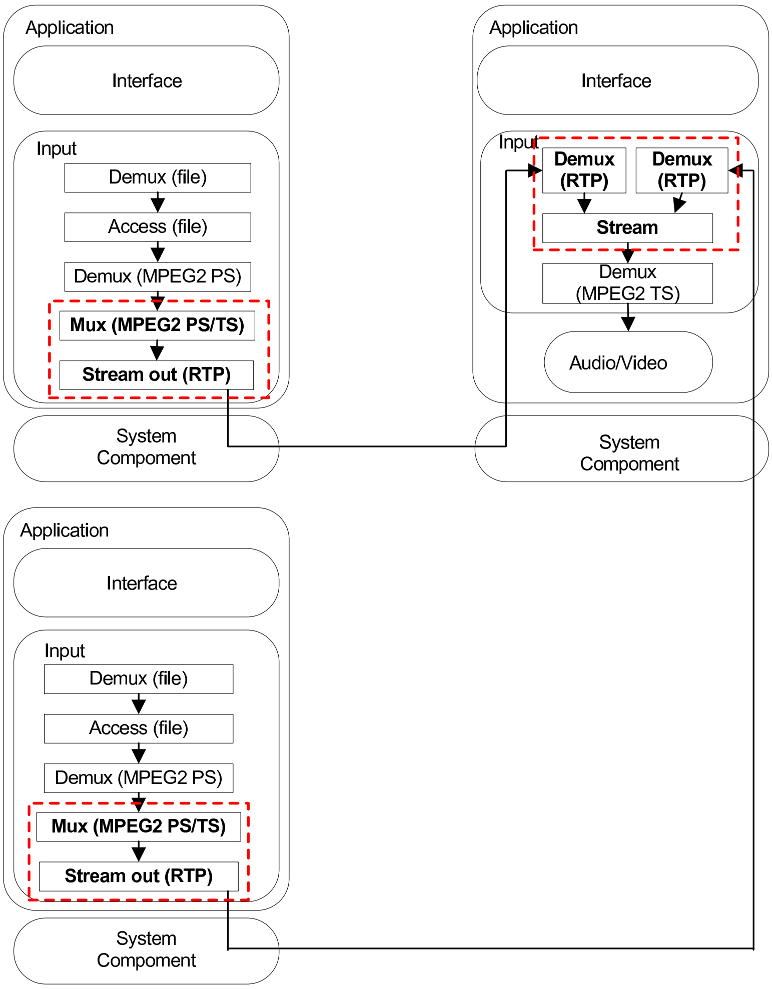



Figure 6.4: VLC module chain for multiple-source based MPEG2 movie streaming

to make sure the servers have started in order to synchronise data transmission from the multiple senders.

The After connection set up, **data scheduling** is required for each of the servers which transmit data. In this thesis, data is equally allocated among the servers. The packet numbering is required in order to support reassembling of streamed data, since there is no reordering mechanism between multiple connections in standard RTP. The RTP header extension [74] is used for packet numbering. Since the extension bit is set, the fixed header is followed by exactly one header extension as presented in Fig 6.5. There is a sequence number field in the RTP header, but that is only for single RTP connection in order to support detecting duplicated or lost packets. The header extension has a simple type, length and value (TLV) structure as shown below. There is no standardised type, so 0x01 is used for distinguishing the packet id. Since the length of packet id field is 4 bytes, the length is set as 1 word.

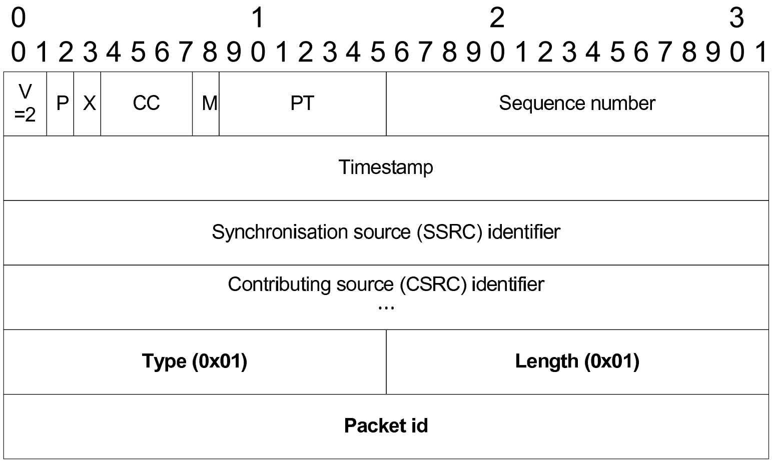



Figure 6.5: Extension enabled RTP header (X=1)

The **sender synchronisation** is required due to timing issue between multiple senders for multiplexing protocols for audio and video for MPEG2 which do not consider multiple-source. MPEG2 TS, which is designed for network delivery, is very dependent on timing since every packet includes a timestamp which is used in decoding. In order to reduce the time difference among senders, the receiver initiated synchronisation between the server when the receiver makes connections. However, MPEG2 PS shows better performance than MPEG2 TS. Therefore, our test results are based on MPEG2 PS in this thesis.

## 6.3 Test Setup Description

The test setup consist of two different configurations for network tests and user perceptual tests, respectively. The network test configuration is based on a wireless network emulation, including a traffic generator. The user perceptual test configuration uses test settings for user perceived quality, which uses the recorded movie clips from network tests.

### 6.3.1 Network Test Setup

The network test configuration includes a conventional network environment connected through WLAN and Wired technologies. A LANforge Traffic Generator [4] is used for emulating a real network environment. The test topology is illustrated in Fig 6.6 and it is emulating the same topology as used in the NS-2 simulation which is based on WLAN which is presented in section 5.3.3. The traffic generator delivers background traffic through wired and wireless environments. The background traffic consists of a number of TCP and UDP flows which change bitrates randomly from 800 Kbps to 1 Mbps. The wireless client PC is equipped with a NETGEAR WG311T wireless card[5] which supports IEEE 802.11g.

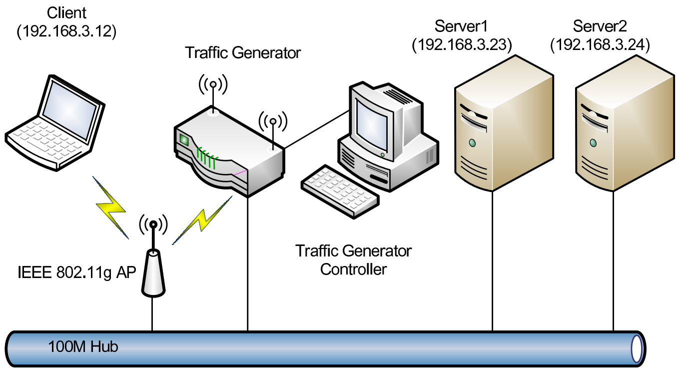



Figure 6.6: Smart PIN prototyping test environment

Using the configuration from Fig. 6.6, servers stream to the client through RTP using different transport protocols such as UDP and DCCP. For the real-life testing, three out of five movies from the simulations are used: "*Die Hard 1*" (DH) - with very high motion content, "*Don't Say A Word*" (DS), with average - low motion content and "*The Road To El Dorado*" (RT) - high motion cartoons. Each movie clip has 3 minutes length involved with multimedia streaming. The transmitted videos are saved for user testing with different testing environments such as streaming approaches and network conditions.

[4]LANforge traffic generator, http://www.candelatech.com/, last accessed 18 Nov. 2009

[5]http://www.netgear.com/Products/Adapters/SuperGWirelessAdapters/WG311T.aspx, last accessed 18 Nov. 2009

### 6.3.2 User Perceived Test Setup

#### 6.3.2.1 Test Methodology

Although very used, PSNR is not a standard method for assessing user perceived quality. Subjective testing is used in order to confirm the results of the objective testing expressed in terms of PSNR. ITU standards, ITU-T P.911 [199], P.910 [200], ITU-R BT-500 [201], etc., are popularly used for measuring subjective quality for specific purposes. However, ITU-T P.911 is good for multimedia system dealing with video and audio, together.

Subjective user testing is performed for multimedia perceptual quality assessment based on ITU-T R. P.911 [199]. As a quality metric, this standard uses Mean Opinion Score (MOS) from the answers to the questions. The users can give a score which is based 5 level quality scale as shown in Table 6.1.

| Rating | Quality | Impairment |
|---|---|---|
| 5 | Excellent | Imperceptible |
| 4 | Good | Perceptible, not annoying |
| 3 | Fair | Slightly annoying |
| 2 | Poor | Annoying |
| 1 | Bad | Very annoying |

Table 6.1: Quality scale for subjective testing (ITU-T R P.911)

There are several criticisms of the ITU-T recommendation for perceived user quality tests that can be found in the literature [202]. In order to comment on those points, the user tests in this thesis include following features:

- ITU P.911 tests require users to watch short sequences of approximately 10 seconds in duration. This is too short for user to assess the quality of multimedia. In this thesis, 30 second clips are used instead.
- The judgements from users are mainly based on the picture quality, but in reality, both audio and video are related to multimedia environment tested. In order to overcome this limitation, the tests include movie clips with video and audio together.

- The test includes only delivered sequences which is degraded. However, the results based on the difference between the degraded and original sequences could be significant. Tests include the original clips from the beginning.
- Perceptual tests in general do not capture change of perception about the quality that users may have during communication under varying network conditions. In order to improve this, the tests include clips delivered through different network conditions.

VLC for Microsoft Windows is the player used for the perceptual test. Test scripts are written using the Perl script language[6]. As shown in Fig. 6.7, the user test room setting includes 4 PC sets with the same monitor and resolution for subjective testing. All the monitors are calibrated using PANTONE huey[7]. The test room is shielded from natural light in order to control noise level and to maintain a constant luminance level. A maximum 4 people could attend the test at the same time. They were informed that they can not have discussions, move equipment and should keep a fixed distance from the monitor. These test setup conform standard recommendation [199].

#### 6.3.2.2 Test Session Description

One session of user tests consists of 23 phases which include short instructions for the flow of the test phase and video for the blind assessment test. Each phase of the test includes three different displays as shown below. The total duration of one phase is about 50 seconds. The questionnaire sheets are available in the Appendix to this thesis.

- Phase title: 5 seconds (e.g. Die Hard 1. Please note clip code (DH) on the questionnaire sheet. Video starts in 5 seconds)
- Movie playing: 30 seconds
- Assessment direction: 15 seconds. 3 questions for each clip. (e.g. please note down your answer on the questionnaire sheet.)

[6]The Perl Directory - perl.org, http://www.perl.org/, last accessed 18 Nov. 2009

[7]Monitor & Printer Profiling - PANTONE huey, http://www.pantone.com/pages/products/product.aspx?pid=79, last accessed 18 Nov. 2009

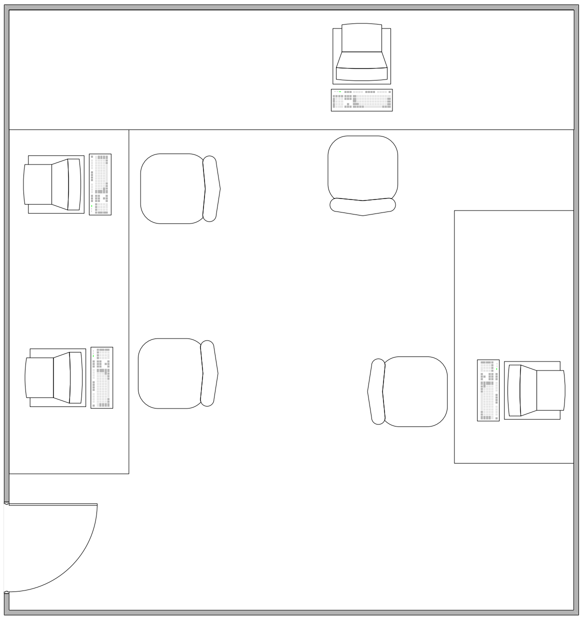


Figure 6.7: Simplified test room environment

As shown in Table 6.2, 30 different movie clips are recorded over network testbed in Fig. 6.6. These clips include 3 different movies which are DH, DS and RT. In addition, 5 streaming schemes are applied. Two different network conditions include low network load and high network load. The low network load includes 3 TCPs and 1 UDP connection and the high network load includes 4 TCPs and 2 UDPs connections. The 5 streaming approaches include Quality Adaptive Multiple-source Multimedia Delivery (QAMMD), Predictive Buffer Algorithm (PBA) [161], a MSDVS-like [85] multiple TFRC connections-based approach (mTFRC) and a PROMISE-like [143] UDP-based multiple streaming solution (mUDP). Single-source based approach (Single) is also used over TFRC protocol, in order to compare with multiple-source streaming approaches.

However, a session of user perceptual test includes only 23 movie clips in order to keep test time less than 30 minutes. These movie clip includes three original movie clips as a preamble. Before starting to view the delivered movie clips, these provide users with some idea of the quality of the original movie clips. The other clips are 20 movie clips from Table 6.2. The total test duration is about 25 minutes. There are three different versions of test set

| Clip id | Approach | Movie | Traffic | Clip id | Approach | Movie | Traffic |
|---|---|---|---|---|---|---|---|
| 01 | Single | DH | Low | 16 | Single | DH | High |
| 02 | mUDP | DS | High | 17 | mUDP | DS | Low |
| 03 | mTFRC | RT | Low | 18 | mTFRC | RT | High |
| 04 | PBA | DH | High | 19 | PBA | DH | Low |
| 05 | QAMMD | DS | Low | 20 | QAMMD | DS | High |
| 06 | Single | RT | High | 21 | Single | RT | Low |
| 07 | mUDP | DH | Low | 22 | mUDP | DH | High |
| 08 | mTFRC | DS | High | 23 | mTFRC | DS | Low |
| 09 | PBA | RT | Low | 24 | PBA | RT | High |
| 10 | QAMMD | DH | High | 25 | QAMMD | DH | Low |
| 11 | Single | DS | Low | 26 | Single | DS | High |
| 12 | mUDP | RT | High | 27 | mUDP | RT | Low |
| 13 | mTFRC | DH | Low | 28 | mTFRC | DH | High |
| 14 | PBA | DS | High | 29 | PBA | DS | Low |
| 15 | QAMMD | RT | Low | 30 | QAMMD | RT | High |

Table 6.2: Movie clip numbering

where each version starts from other clips for 20 movie clips. For example, test set 1 uses from clip id 01 to 20, test set 2 uses from clip id 11 to 31, and test set 3 uses from clip id 21 to 10.

## 6.4 Network Performance Test and Results Analysis

Network test results include Peak Signal to Noise Ratio (PSNR) and initial waiting time. In order to measure PSNR, the MSU Video Quality Measurement Tool[8] is used. However, this tool has a limitation of quality measurement for streamed video. The tool provides result visualisations as shown in Fig. 6.8. During measurements, quality degradation is related to data loss in a frame. Fig. 6.9 shows the frame which has errors. In addition, the tool considers that there is a quality degradation when a frame is not synchronised. However, the actual video quality is not dropped significantly with out-of-sync frames as shown in Fig. 6.10. In order to reduce synchronisation issue, the first 10 seconds of PSNR measurements are used in this thesis. The total waiting time is not considered for prototyping tests in this

[8]MSU Video Quality Measurement Tool (PSNR, MSE, VQM, SSIM), http://compression.ru/video/quality_measure/video_measurement_tool_en.html, last accessed 18 Nov. 2009

thesis since VLC does not support re-buffering. In total, 10 tests are involved for every scheme and movie clip.

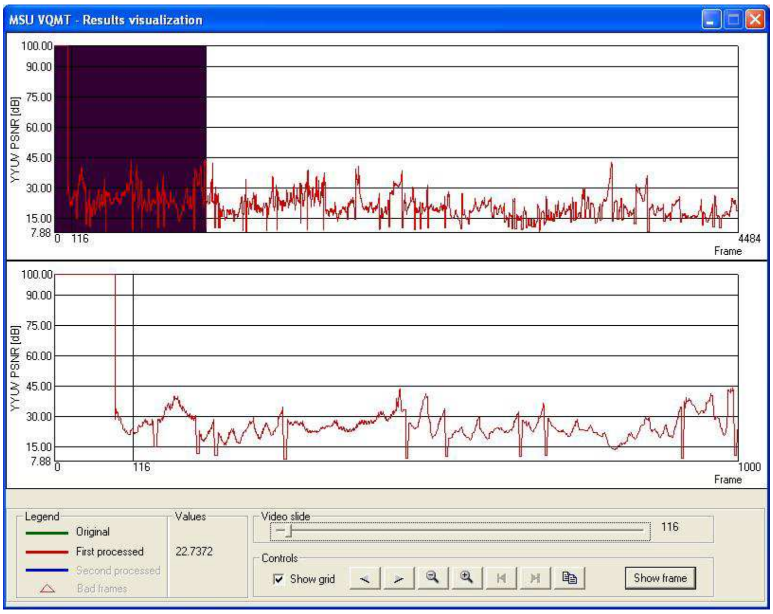



Figure 6.8: PSNR result visualisation


(a) Source

(b) Destination

Figure 6.9: An example of an error frame

Table 6.3 and Fig. 6.11 show a comparison between schemes in terms of PSNR under different network conditions. The low traffic condition has 4 background connections including 3 TCPs and 1 UDP. The high traffic condition has 6 background connections including 4 TCPs and 2 UDPs. A TCP or UDP connection changes bitrates randomly from


(a) Source (b) Destination

Figure 6.10: An example of out-of-sync frame

800 Kbps to 1 Mbps, as mentioned previously.

On average, the TFRC-based approaches such as QAMMD, mTFRC and PBA show high PSNR values. QAMMD has 92.9 dB whereas mTFRC and PBA have 87.0 dB and 87.8 dB, respectively. It can be seen how QAMMD behaves with 5.8% better than PBA, with 6.8% better than mTFRC. However, very low PSNR values are achieved as 41.0 dB and 16.8 dB when Single and mUDP are employed. When high traffic is found, QAMMD shows greater benefit than other approaches. QAMMD has 93.3 dB whereas mTFRC and PBA have 81.3 dB and 81.8 dB, respectively. It can be seen how QAMMD behaves with 14.1% better than PBA, with 14.8% better than mTFRC. In the case of low traffic, TFRC based approaches such as QAMMD, mTFRC and PBA show similar PSNR values. QAMMD has 92.6 dB whereas mTFRC and PBA have 94.3 dB and 92.3 dB, respectively.

The average initial waiting time is presented in Table 6.4. PBA and QAMMD are compared since only those adopt any kind of buffering algorithm. On average, when using QAMMD, the initial waiting time is 44.7 seconds, whereas when PBA is employed the initial waiting time is 41.0 seconds.

| | | Single | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|---|---|
| 3 TCPs 1 UDP | DH | 37.8 | 17.0 | 94.0 | 90.0 | 89.7 |
| | DS | 50.2 | 19.7 | 98.0 | 96.9 | 95.3 |
| | RT | 47.2 | 13.8 | 84.9 | 96.1 | 92.7 |
| | Average | 45.1 | 16.8 | 92.3 | 94.3 | 92.6 |
| 4 TCPs 2 UDPs | DH | 34.8 | 16.9 | 79.8 | 82.2 | 96.5 |
| | DS | 36.7 | 20.1 | 76.8 | 77.7 | 94.3 |
| | RT | 39.1 | 13.5 | 88.8 | 84.1 | 89.2 |
| | Average | 36.9 | 16.9 | 81.8 | 81.3 | 93.3 |
| Total average | | 41.0 | 16.8 | 87.0 | 87.8 | 92.9 |

Table 6.3: Measured PSNR statistics (dB)

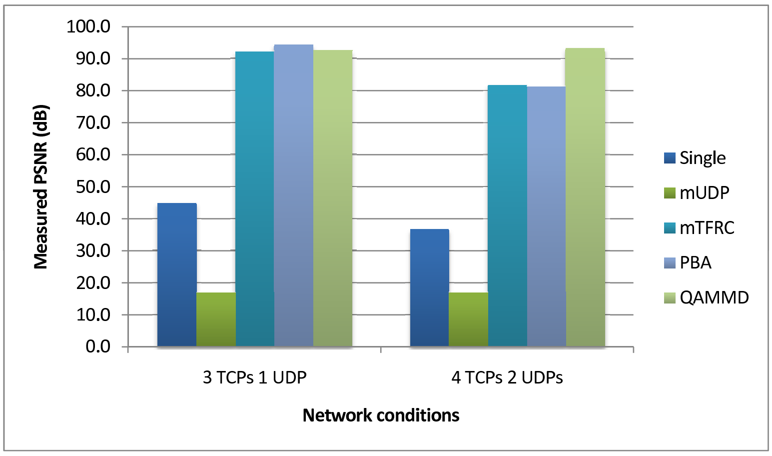



Figure 6.11: Measured PSNR statistics chart

| | | PBA | QAMMD |
|---|---|---|---|
| 3 TCPs 1 UDP | DH | 40.8 | 46.1 |
| | DS | 42.5 | 46.2 |
| | RT | 38.3 | 44.3 |
| | Average | 40.6 | 45.6 |
| 4 TCPs 2 UDPs | DH | 42.3 | 46.1 |
| | DS | 41.0 | 43.8 |
| | RT | 37.7 | 41.9 |
| | Average | 41.0 | 44.7 |
| Total average | | 40.5 | 45.1 |

Table 6.4: Initial waiting time statistics (secs)

## 6.5 User Perceived Quality Test and Results Analysis

The user perceptual test includes 3 questions which are related to quality, continuity and synchronisation between the video and audio from a multimedia clip. The continuity of the video is adopted for another aspect of the quality of the movie. Since there are multiple-source approaches, user perception of synchronisation is also included as one of the questions. Based on ITU-T R. P.911 [199], Mean Opinion Score (MOS) is used as it is mentioned. The following tables present the MOS for each answer from 30 users who are on average 27.5 years old students. The test result includes a longer duration for the assessments since each phase has 30 seconds of playing time.

Table 6.5 and Fig. 6.12 show a comparison between schemes in terms of MOS on the quality of multimedia clips with different network conditions. On average, when using QAMMD, MOS is 3.7, whereas when Single, mUDP, mTFRC and PBA are employed, MOS values are 3.1, 1.4, 2.6 and 3.3, respectively. It can be seen how QAMMD behaves with 12.1% better performance than PBA, with 42.3% better performance than mTFRC and with 19.4% better performance than Single. In the case of mUDP, QAMMD shows about 1.6 times better MOS. In addition, QAMMD has higher MOS than PBA using t-test ($t = 1.76$, $d.f. = 5$, $p < 0.07$). Specifically, when the wireless bottleneck channel becomes crowded with 4 TCPs and 2 UDPs, QAMMD offers 32.3% better perceived quality than PBA, 70.8% better perceived quality than mTFRC and 32.3% better than Single, as expressed in terms of MOS for quality. In the case of mUDP, QAMMD shows about 2.2 times better MOS.

Table 6.6 and Fig. 6.13 show a comparison between schemes in terms of MOS on the video continuity of the multimedia clips under different network conditions. The overall results are similar to MOS for quality. On average, when using QAMMD, MOS is 3.8, whereas when Single, mUDP, mTFRC and PBA are employed, MOS is 3.1, 1.2, 2.5 and 3.2, respectively. It can be seen that QAMMD behaves with 18.8% better performance than PBA, with 52.0% better performance than mTFRC and with 22.6% better performance than Single. In the case mUDP, QAMMD shows about 2.2 times better MOS. In addition,

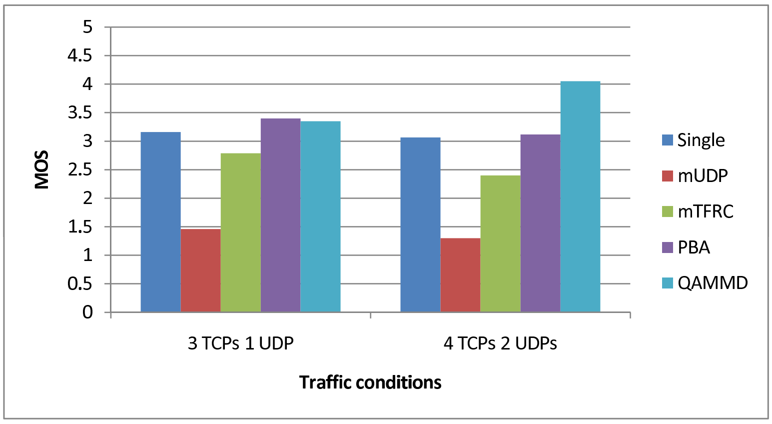



Figure 6.12: MOS for quality of multimedia clip

| | | Single | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|---|---|
| 3 TCPs 1 UDP | DH | 2.3 | 1.5 | 3.3 | 3.5 | 3.5 |
| | DS | 3.6 | 1.4 | 3.4 | 3.3 | 3.1 |
| | RT | 3.6 | 1.5 | 1.8 | 3.5 | 3.5 |
| | Average | 3.2 | 1.5 | 2.8 | 3.4 | 3.4 |
| 4 TCPs 2 UDPs | DH | 2.8 | 1.4 | 3.3 | 2.3 | 3.8 |
| | DS | 3.3 | 1.1 | 1.9 | 3.5 | 4.3 |
| | RT | 3.1 | 1.4 | 2.1 | 3.5 | 4.1 |
| | Average | 3.1 | 1.3 | 2.4 | 3.1 | 4.1 |
| Total average | | 3.1 | 1.4 | 2.6 | 3.3 | 3.7 |

Table 6.5: Mean Opinion Score for quality

QAMMD has higher MOS than PBA using t-test ($t = 1.55$, $d.f. = 5$, $p < 0.1$). Specifically, when the wireless bottleneck channel is crowded with 4 TCPs and 2 UDPs, QAMMD offers 57.1% better perceived quality than PBA, 95.5% better perceived quality than mTFRC and 34.4% better than Single, as expressed in terms of MOS for quality. In the case of mUDP, QAMMD shows about 2.6 times better MOS.

Table 6.7 and Fig. 6.14 show a comparison between schemes in terms of MOS on the synchronisation between video and audio of multimedia clip under different network conditions. The overall result is yet again similar to MOS for quality although multiple-source streaming is involved. On average, when using QAMMD, MOS is 4.0, whereas when Single, mUDP, mTFRC and PBA are employed, MOS is 3.7, 1.9, 3.2 and 3.8, respectively. Once again it can be seen that QAMMD behaves with 5.3% better performance than PBA,

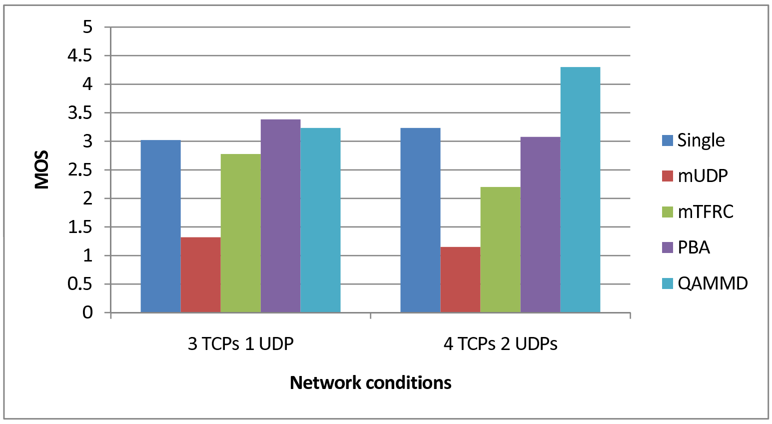



Figure 6.13: MOS for continuity of video

| | | Single | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|---|---|
| 3 TCPs 1 UDP | DH | 2.2 | 1.5 | 3.4 | 3.8 | 3.1 |
| | DS | 3.5 | 1.3 | 3.6 | 2.8 | 3.2 |
| | RT | 3.4 | 1.2 | 1.3 | 3.7 | 3.5 |
| | Average | 3.0 | 1.3 | 2.8 | 3.4 | 3.2 |
| 4 TCPs 2 UDPs | DH | 3.4 | 1.4 | 3.0 | 2.6 | 4.1 |
| | DS | 2.9 | 1.1 | 2.0 | 3.5 | 4.7 |
| | RT | 3.4 | 1.1 | 1.7 | 3.2 | 4.1 |
| | Average | 3.2 | 1.2 | 2.2 | 3.1 | 4.3 |
| Total average | | 3.1 | 1.2 | 2.5 | 3.2 | 3.8 |

Table 6.6: Mean Opinion Score for continuity of video

with 25.0% better performance than mTFRC and with 8.1% better performance than Single. In the case mUDP, QAMMD shows about 1.1 times better MOS. In addition, QAMMD has higher MOS than PBA using t-test ($t = 1.30$, $d.f. = 5$, $p < 0.13$). Specifically, when the wireless bottleneck channel is crowded with 4 TCPs and 2 UDPs, QAMMD offers 19.4% better perceived quality than PBA, 43.3% better perceived quality than mTFRC and 13.2% better than Single expressed in terms of MOS for quality. In the case of mUDP, QAMMD shows about 1.4 times better MOS.

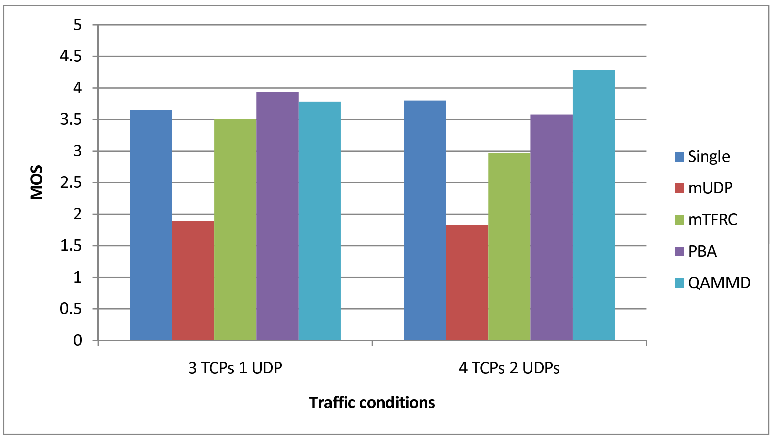



Figure 6.14: MOS for synchronisation between video and audio

| | | Single | mUDP | mTFRC | PBA | QAMMD |
|---|---|---|---|---|---|---|
| 3 TCPs 1 UDP | DH | 3.1 | 2.4 | 4.0 | 4.1 | 3.8 |
| | DS | 4.0 | 1.6 | 4.3 | 3.9 | 3.7 |
| | RT | 3.9 | 1.7 | 2.3 | 3.9 | 3.9 |
| | Average | 3.6 | 1.9 | 3.5 | 3.9 | 3.8 |
| 4 TCPs 2 UDPs | DH | 3.9 | 1.9 | 3.7 | 3.1 | 4.1 |
| | DS | 4.0 | 1.7 | 2.9 | 3.8 | 4.7 |
| | RT | 3.6 | 2.0 | 2.4 | 3.8 | 4.1 |
| | Average | 3.8 | 1.8 | 3.0 | 3.6 | 4.3 |
| Total average | | 3.7 | 1.9 | 3.2 | 3.8 | 4.0 |

Table 6.7: Mean Opinion Score for synchronisation between video and audio

## 6.6 Comparison with Simulation Test

The prototyping test includes similar test settings to wireless dumbbell simulation. The simulation test results are discussed in detail in section 5.4.3.3. Prototyping test results are divided into PSNR measurements and subjective user perceived quality assessment.

In terms of PSNR, the simulation test shows that QAMMD offers 42.3% better perceived quality than PBA and 90.7% better perceived quality than mTFRC with 4 nodes which make network highly loaded. During prototyping, it can be seen how QAMMD behaves with 14.1% better than PBA, with 14.8% better than mTFRC. Due to difference of tests such as the PSNR measurement duration, performance benefit may vary. However, QAMMD provides better performance than other approaches.

As mentioned, QAMMD behaves with 12.1% better performance than PBA and with 42.3% better performance than mTFRC in terms of quality based on subjective test. Similar to comparison of simulation test and prototyping test, QAMMD has more benefit than other approaches.

## 6.7 Conclusion

This chapter included a description of the prototyping of Smart PIN which is mainly focused on Quality Adaptive Multiple-source Multimedia Delivery (QAMMD) using comparison-based tests. The metrics used include Peak Signal to Noise Ratio (PSNR) as an objective metric and Mean Opinion Score (MOS) as a subjective metric for quality of multimedia delivered to the user. In terms of PSNR, QAMMD shows better performance in comparison to other schemes, especially for cases where there is more traffic over the wireless network. In the next chapter, the conclusion of the thesis and future work are presented.

# Chapter 7

# Conclusions and Future Work

## 7.1 Conclusions

Information overload is becoming a fundamental problem in modern society as we struggle to keep track of all the information available to us. In addition, we find that the generation of digital content is increasingly including pictures, video and audio sequences. Furthermore, users do not only consume content from within a single device, or not even from a set of their own devices, but they want to share digital content with other users as well, and they want to do this in real time and naturally without any loss of qualty. Although evolving networking technologies enable content sharing, the delivery of content is still cost-dependent and is limited by network bandwidth in today's environments which have heterogeneous network technologies. However, these are just technical issues and what users want is simple and efficient usage and exchange of their content in spite of these network delivery-related difficulties.

This thesis presented Smart PIN, a novel performance and cost- oriented context-aware Personal Information Network. Smart PIN's architecture includes network, service and management components. Within the service component, two novel schemes for efficient delivery of context and content data were proposed and developed: a Multimedia Data Replication Scheme (MDRS) and a Quality-oriented Algorithm for Multiple-source Multimedia Delivery(QAMMD). MDRS targets efficient user access to information located on remotely distributed devices in a heterogeneous network environment. QAMMD employs buffer underflow avoidance scheme (BUAS) using a receiving buffer in the receiver in order to maintain quality at high levels under varying network conditions during multimedia delivery.

Assessment of the performance of the MDRS scheme is presented in terms of both modelling and simulations, and of a prototype implementation. Modelling and simulations are performed using Network Simulator 2. The Smart PIN model employed during simulations includes not only the service component and applications, but also the Media Independent Handover (MIH) as a virtually always-connected heterogeneous network model.

The proposed MDRS is compared with two other autonomous well-known data replication schemes based on data availability and network resource usage control: Autonomous Replication (AR) and Tempo. MDRS shows better performance in terms of data availability and network resource usage especially with information overload and segmented data sharing. In terms of multiple-source streaming, the proposed QAMMD is compared with a multiple TFRC connection-based scheme (mTFRC) and a multiple UDP connection based scheme (mUDP). In addition, a similar buffer estimation algorithm, the Predictive Buffer Algorithm (PBA) is also used in the comparisons. Performance is evaluated for multiple-source streaming approaches in terms of the estimated Peak Signal to Noise Ratio (PSNR) and overhead measurements. QAMMD shows better performance in terms of PSNR using different network topologies including wired, WLAN and WLAN/WiMAX topologies. In addition, it shows less buffer underflow and less overall waiting time.

Both simulation and prototyping-based tests also assessed the performance of the proposed QAMMD scheme. They use PSNR as an objective metric and Mean Opinion Score (MOS) as a subjective metric for quality of multimedia. In terms of PSNR, QAMMD shows better performance compared with other schemes such as mTFRC, mUDP, PBA, and single source streaming, especially when there is more traffic over the wireless network. In addition, a set of user perceptual tests which we performed shows that QAMMD presents better quality multimedia than other similar approaches.

The thesis was structured as follows. After the description of the motivation behind the work, the problem addressed, the solution proposed and the major contributions were all then presented in the Introduction. Chapter 2 presented the various background solutions related to networking technologies, context and metadata, data replication and transport protocols, which had already been available. Chapter 3 included a detailed literature review of the various research aspects in relation to Next Generation Networks (NGNs), person-centric context-aware systems, and data replication systems. Furthermore, realtime applications for multimedia delivery were discussed as well as non-realtime applications in terms of similar approaches to Smart PIN.

In chapter 4, the Smart PIN system architecture was presented from both points of view: network-orientation and context-aware personal information system. The novel Multimedia Data Replication Scheme (MDRS) and the Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD) were also introduced and then presented in detail in this chapter.

Chapter 5 included the comparison-based simulation tests involving Smart PIN and the two associated algorithms: MDRS and QAMMD and showed the benefits of using Smart PIN in terms of user perceived quality and overhead.

Chapter 6 included a description of the prototyping-based tests involving Smart PIN, which were mainly focused on QAMMD. Different multimedia clips and a variety of network conditions were considered, and the comparison-based tests showed how the proposed QAMMD was better than other existing approaches.

## 7.2 Contributions

The principal contributions of this thesis are the Multimedia Data Replication Scheme (MDRS) and a Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD). In order to combine these two, Smart PIN also requires having a new context-aware personal information system design which supports three different services as a context-aware system. MDRS is based on a utility function which supports cost, user interest and content popularity. MDRS also includes minimum data set requirements for segmented data. The results of tests performed show that the proposed MDRS outperforms other general purpose data replication schemes such as AR and Tempo. MDRS uses a utility function for data selection and delivery which enables us to achieve better data availability and network usage. QAMMD is a multiple-source streaming approach which adopts a novel buffer underflow avoidance scheme (BUAS) supporting high multimedia quality. QAMMD provides better quality related performance when it is compared agaionst other streaming schemes using simulation and prototyping.

## 7.3 Future Work

In terms of next generation network, user-centric approaches can be a major research area since the most approaches consider issues below the transport layer and there is a lack of consideration for the application level. In addition, there are still missing attributes from the view of combining user, content, device and network issues. Up to now, most approaches only consider one or two aspects such as type of content and network (i.e. IEEE 802.11e), content and device profile (i.e. MPEG-21 content adaptation), networks and devices (i.e. NGN approaches). The consideration of heterogeneity of these four aspects provide many challenges in order to maintain high levels of user's perceived quality of multimedia services, etc. Due to this situation, similar approaches to Smart PIN will get more popular. As future work, two extending approaches for Smart PIN are introduced, as well as two long-term approaches are discussed.

The work described in this thesis is mainly focused on network-oriented aspects for MDRS and QAMMDS. As a extended work of MDRS, context annotation could be a topic for future work for Smart PIN research. MDRS can be adopted in a content-sharing system which supports a content search or filtering feature in order to allow the user to access content more easily. During this procedure, there could be useful contextual information such as users, their location, time, popularity of content, etc. This could be also stored as metadata and thus part of the content and used for content sharing efficiently as another prototyping approach. One good starting point is Beagle[1] which provides personalised search function on Linux-based PC. Using the file indexing feature of Beagle, the proposed categorisation and delivery could be implemented. In addition, the effect of changing the parameters dynamically by the user by, for example, changing the popularity rating, could be examined including utility thresholds and device availability.

Another extension for Smart PIN could be the QAMMD optimisation. QAMMD includes initial delay for buffer estimation, and there is no adaptation after starting playout. These two part can be enhanced. During the initial delay, there could be some approach to provide higher user satisfaction such as gradual display or reducing delays. Second approach could include adaptation approaches using play buffer during playout. One of possible way to do adaptation is adjustment of time duration between frames [203]. Usually, duration between video frames is fixed, but users hardly recognise if the duration is changed in certain range. This kind of approach is called as Adaptive Multimedia Playout. Additionally, other adaptive delivery approaches can be employed to cope with the adaptation during playout due to changing network delivery conditions.

As a long term approach, Smart PIN relaxes assumptions made by other approaches. An assumption of Smart PIN is that it uses an overlay network similar to BitTorrent[2]. Distributed Indexing techniques such as Distributed Hash Tables (DHT) can be used for Smart PIN. Currently, data replication with DHT is only used to achieve better data availability. This can be extended to achieve higher users utility. MDRS already supports this characteristic, but it is still a challenge because of the dynamics of DHT. Similar approach to Plover [119] can be used in order to resolve the related issues, but there still additional challenges in order to support efficient data delivery such as QAMMD.

[1]Main Page - Beagle, http://beagle-project.org/Main_Page, last accessed 18 Nov. 2009

[2]The BitTorrent Protocol Specification, http://www.bittorrent.org/beps/bep_0003.html, last accessed 18 Nov. 2009

Media Independent Handover (MIH) is used as a network component in this thesis. However, there could be more efficient ways to achieve better performance on cost-dependent networks in terms of routing and search algorithms. Specifically, the management component can be involved in order to achieve balanced traffic control between background and foreground traffic. As candidate approach, admission controls such as [204] are interesting avenues. Based on the admission control, background and foreground traffic of a user could be managed efficiently as well as traffic among users.

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# Appendices

## Appendix A: PSNR estimation

From the definition, PSNR is presented as Eq. 7.1 where $MAX$ is maximum possible pixel value and $MSE$ is mean square error.

$$PSNR = 20 \cdot log_{10}\left(\frac{MAX}{\sqrt{MSE}}\right) \tag{7.1}$$

In order to apply to network delivery, the average bitrate of the multimedia stream after the encoding process($MAX\_Bitrate$) is, the average throughput expected when adaptively delivering the multimedia stream over the network ($EXP\_Thr$), and the actual throughput measured during delivery ($CRT\_Thr$) are introduced.

The $MAX$ is defined as total data need to receive in Eq. 7.2 where $T$ is duration of multimedia clip.

$$MAX = MAX\_Bitrate \cdot T \tag{7.2}$$

Due to the loss of delivery of data, Mean square error, $MSE$ is defined using $EXP\_Thr$ and $CRT\_Thr$ in Eq. 7.3.

$$\begin{aligned} MSE &= (EXP\_Thr \cdot T - CRT\_Thr \cdot T)^2 \\ &= T^2 \cdot (EXP\_Thr - CRT\_Thr)^2 \end{aligned} \tag{7.3}$$

With Eq. 7.1, Eq. 7.2 and Eq. 7.3, $PSNR_{EST}$ is derived as Eq. 7.4.

$$
\begin{aligned}
PSNR_{EST} &= 20 \cdot log_{10}\left(\frac{MAX\_Bitrate \cdot T}{\sqrt{T^2 \cdot (EXP\_Thr - CRT\_Thr)^2}}\right) \\
&= 20 \cdot log_{10}\left(\frac{MAX\_Bitrate}{\sqrt{(EXP\_Thr - CRT\_Thr)^2}}\right)
\end{aligned}
\tag{7.4}
$$

# Appendix B: Alternative Minimum Data Set Decision

In order to enhance $G^k$ which is introduced in section 4.3.4, $G_2^k$ is introduced based on the definition of data availability as in Eq. 7.5 where $D^k_{Target,l}$ is target data availability and $J$ is the total number devices which have segment $l$ of multimedia data item $k$.

$$D^k_{Target,l} = 1 - \prod_{i=1}^{J}(1 - P_i) \tag{7.5}$$

In order to simplify estimation of Eq. 7.5, $P_{avg}$ can be used instead of $P_i$ as Eq. 7.6.

$$1 - D^k_{Target,l} = (1 - P_{avg})^J \tag{7.6}$$

Taking $\log$ both side of Eq. 7.6, $J$ can be presented as Eq. 7.7.

$$\log(1 - D^k_{Target,l}) = J \cdot \log(1 - P_{avg}) \tag{7.7}$$

Denoting that $J$ is also considered as number of data, $G_2^k$ which could be derived using Eq. 7.8.

$$G_2^k = \left\lceil \log_{(1-P_{avg})}(1 - D^k_{Target,l}) \right\rceil \tag{7.8}$$

Fig. 7.1 presents minimum data set against data availability using $G^k$ and $G_2^k$,where $D^k_{Target,l}$ is 0.995. $G_2^k$ shows more data set is required for smaller device availability than $G^k$. However, the proposed Multimedia Data Replication Scheme has less impact since data delivery does not related to this aspect. However, the time for acquiring this much data set should be increased.

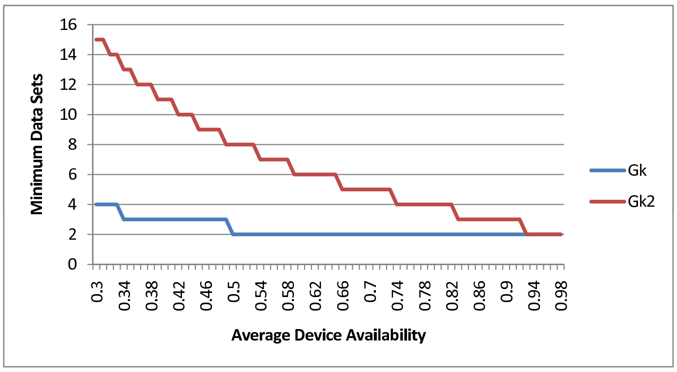


Figure 7.1: Minimum data set graph against average data availability

# Appendix C: Abbreviations

AAS: Advanced Antenna System

ADSL: Asynchronous Digital Subscriber Line

AIMDL: Additive Increase/Multiplicative Decrease

ALM: Application Level Multicast

AODV: Ad-hoc On-Demand Distance Vector Routing

AON: Active Optical Network

AP: Access Point

ARQ: Automatic Repeated reQuest

ASPAN: Auto-configuration and Self-management of Personal Area Networks

ATIS: Alliance for Telecommunications Industry Solutions

BCM: Buffer Coordination Module

BSS: Basic Service Set

BUAS: Buffer Underflow Avoidance Scheme

BWA: Broadband Wireless Access

CCID: Congestion Control Identifiers

CDN: Content Distribution Network

CM: Cable Modem

CMTS: Cable Modem Termination System

CSMA/CD: Carrier Sense Multiple Access Collision Detect

DAMA: Demand Assigned Multiple Access

DCCP: Datagram Congestion Control Protocol

DCF: Distributed Coordination Function

DFS: dynamic frequency selection

DHT: Distributed Hash Table

DIFS: Distributed Inter Frame Space

DSDV: Destination-Sequenced Distance Vector routing

DSL: Digital Subscriber Line

DSLAM: Digital Subscriber Line Access Multiplexer

DSSS: Direct Sequence Spread Spectrum

DVB: Digital Video Broadcasting

ESS: Extended Service Set

FIFO: First In First Out

FDD: Frequency Domain Duplexing

FDM: Frequency Domain Multiplexing

FHSS: Frequency Hopping Spread Spectrum

FTTH: Fiber to the Home)

FMCA: Fixed-Mobile Convergence Alliance

GENA: Generic Event Notification Architecture

GSM: Global System for Mobile Communications

HG: Home Gateway

IBSS: Independent Basic Service Set

IMS: IP Multimedia Subsystem

IP: Internal Protocol

IPTV: Internal Protocol Television

ITU-T: International Telecommunication Union Telecommunication Standardisation Sector

LAN: Local Area Network

LLC: Logical Link Control

MAC: Medium Access Control

MDC: Multiple-Description Coding

MDRS: Multimedia Data Replication Scheme

MIH: Media Independent Handover

MIMO: Multi-Input, Multiple Output

MPEG2: Moving Picture Expert Group 2

MPEG2 PS: Moving Picture Expert Group 2 Program Stream

MPEG2 TS: Moving Picture Expert Group 2 Transport Stream

MPEG4 FGS: MPEG4 Fine-Granularity Scalable

MP-MP: MultiPoint-to-MultiPoint

ND: Neighbour Discovery

NGN: Next Generation Network

OFDM: Orthogonal Frequency-Division Multiplexing

OSI: Open System Interconnection

PCF: Point Coordination Function

PVR: Personal Video Recorder

PON: Passive Optical Network

QAMMD: Quality-oriented Algorithm for Multiple-source Multimedia Delivery

RTCP: Real-time Transport Control Protocol

RTP: Real Time Protocol

SNR: Signal to Noise Ratio

PBA: Predictive Buffering Algorithm

PMP: point-to-multipoint

PSNR: Peak Signal to Noise Ratio

QoS: Quality of Service

RAP: Rate Adaptation Protocol

RTCP: Real Time Control Protocol

RTP: Real Time Protocol

RTSP: Real Time Streaming Protocol

SHE: Super Head End

SDP: Service Discovery Protocol

SIP: Session Initiation Protocol

Smart PIN: Smart Personal Information Network

SOAP: Simple Object Access Protocol

SSDP: Simple Service Discovery Protocol

SSON: Service Specific Overlay Network

STB: Set Top Box

STC: Space Time Coding

TCP: Transmission Control Protocol

TDD: Time Division Duplexing

TDM: time division multiplexing

TDMA: Time Division Multiple Access

TFRC: TCP Friendly Rate Control

TLV: Type, Length and Value

UMA: Unlicensed Mobile Access

UMTS: Universal Mobile Telecommunication System

UPnP: Universal Plug and Play

UPnP AV: UPnP Audio Video

URI: Uniform Resource Identifier

USB: Universal Serial Bus

VLC: Video LAN Client

VSO: Video Service Office

UDP: User Datagram Protocol

UMTS: Universal Mobile Telecommunications System

UP2P: Ubiquitous P2P

VHE: Virtual Home Environment

VHO: Video Hub Office

VoD: Video on Demand

VSO: Video Service Office

WAMS: Wide-Area Media Sharing

WLAN: Wireless Local Area Network

WMAN: Wireless Metropolitan Area Network

WPAN: Wireless Personal Area Network

XML: eXtended Markup Language

xUPnP: eXtended UPnP

# Appendix C: Test material

## Test Instructions

**Welcome Message**
Welcome to the perceptual testing session organised by Dublin City University.

**Test Objectives**
We are trying to figure out the relationship between different multimedia streaming schemes. This subjective perceptual test will be utilized for background knowledge of new strategy of multimedia streaming.

**Disclaimer**
Please fill in the personal information page. The information collected will be utilized as reference for analysing the perceptual test results and will never be made public in any form**.**

**Test Directions**
The test set number will be displayed at the beginning. Please write down that test set number on the questionnaire sheet. The test consists of eleven phases. In each phase you will be shown a test clip and you will be asked to grade its quality on the indicated 1-5 scale, where 1 is the worst quality ("bad") and 5 - the best ("excellent"). The grading is done immediately after the clip ended. You are not allowed to change the screen position, the distance from the screen since they are fixed for all the test subjects. Once the test has started you are not allowed to pause it or to stop it or to ask questions. The test will take less than 30 minutes.

## Personal Information Page

Please, check with "√" for your choice.

| | |
|---|---|
| **Record No:** | |

| | | | |
|---|---|---|---|
| **Gender:** | **a) male** | **b) female** | |

| | |
|---|---|
| **Age:** | |

| | | | |
|---|---|---|---|
| **Do you use glasses/contact lenses:** | **a) yes** | **b) no** | |

| | | | |
|---|---|---|---|
| **Are you long/short sighted:** | **a) long sighted** | **b) short sighted** | **c) no** |

| | | | |
|---|---|---|---|
| **Do you have other visual conditions that may affect your perception of movies (e.g. colour blindness, glare):** | **a) yes** | **b) no** | |

| | | | |
|---|---|---|---|
| **How familiar are you with multimedia streaming:** | **a) I work in this domain** | **b) I am familiar** | **c) I am not familiar** |

| | | | |
|---|---|---|---|
| **Dou you rent DVDs/tapes:** | **a) often** | **b) sometimes** | **c) never** |

| | | | |
|---|---|---|---|
| **Do you go to cinema/theatre:** | **a) often** | **b) sometimes** | **c) never** |

| | | | |
|---|---|---|---|
| **How long are you willing to wait for buffering when you use the Internet to watch high quality movie (e.g. DVD quality) :** | **a) < 10 secs** | **b) 30 sec** | **c) > 50 secs** |

# Questionnaire

| Record No: | |
|---|---|

| Test Type: | |
|---|---|

**Directions**

When the test is started, you can see the test set number such as “T1”. There are 23 phases in the test. On each phase you will see phase name abbreviation such as “DH”, “DS” and such a things on the screen. **And, could you kindly answer the following questions about each sequence shown?**

A) Give a tick under the perceived quality of the streamed movie clip from 1 (the worst quality) to 5 (the best) subjective scales presented in the following table.
B) Give a tick under the description on the continuity of the movie clip from 1 (too jerky) to 5 (smooth) subjective scales.
C) Give a tick under the description on the synchronisation between video and audio of the movie clip from 1 (bad) to 5 (excellent) subjective scales.

**Quality scale for subjective testing (ITU-T R P.911)**

| **Rating** | **Quality** | **Impairment** |
|---|---|---|
| **5** | **Excellent** | **Imperceptible** |
| **4** | **Good** | **Perceptible, not annoying** |
| **3** | **Fair** | **Slightly annoying** |
| **2** | **Poor** | **Annoying** |
| **1** | **Bad** | **Very annoying** |

**Example for a phase of Answer Sheet**

| **Phase No:** 4 |
|---|

**A) Tick the perceived quality of the multimedia clip:**

| **5. Excellent** | **4. Good** | **3. Fair** | **2. Poor** | **1. Bad** |
|---|---|---|---|---|
| | √ | | | |

**B) Tick what you think on the continuity of the video clip:**

| **5. Smooth** | **4** | **3. Somewhat jerky** | **2** | **1. Too jerky** |
|---|---|---|---|---|
| | | √ | | |

**C) Thick what you think on the synchronisation between video and audio:**

| **5. Excellent** | **4. Good** | **3. Fair** | **2. Poor** | **1. Bad** |
|---|---|---|---|---|
| | | | √ | |

# Answer Sheet

| Test Set No: | |
|---|---|

| Phase No: 1 |
|---|

A) Tick the perceived quality of the multimedia clip:

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) Tick what you feel on the continuity of the video clip:

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) Thick what you think on the synchronisation between video and audio:

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: 2 |
|---|

A) Tick the perceived quality of the multimedia clip:

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) Tick what you feel on the continuity of the video clip:

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) Thick what you think on the synchronisation between video and audio:

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: 3 |
|---|

A) Tick the perceived quality of the multimedia clip:

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) Tick what you feel on the continuity of the video clip:

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) Thick what you think on the synchronisation between video and audio:

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 4 |
|---|---|

**A) Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

**B) Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

**C) Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 5 |
|---|---|

**A) Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

**B) Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

**C) Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 6 |
|---|---|

**A) Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

**B) Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

**C) Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 7 |
|---|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 8 |
|---|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 9 |
|---|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 10 |
|---|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 11 |
|---|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 12 |
|---|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 13 |
|---|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 14 |
|---|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 15 |
|---|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: 16 |
|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: 17 |
|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: 18 |
|---|

A) **Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

B) **Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

C) **Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 19 |
|---|---|

**A) Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

**B) Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

**C) Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 20 |
|---|---|

**A) Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

**B) Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

**C) Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

| Phase No: | 21 |
|---|---|

**A) Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

**B) Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

**C) Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

**Phase No: 22**

**A) Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

**B) Tick what you feel on the continuity of the video clip:**

| 5. Smooth | 4 | 3. Somewhat jerky | 2 | 1. Too jerky |
|---|---|---|---|---|
| | | | | |

**C) Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

**Phase No: 23**

**A) Tick the perceived quality of the multimedia clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

**B) Tick what you feel on the continuity of the video clip:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |

**C) Thick what you think on the synchronisation between video and audio:**

| 5. Excellent | 4. Good | 3. Fair | 2. Poor | 1. Bad |
|---|---|---|---|---|
| | | | | |