DOKUZ EYLÜL UNIVERSITY
GRADUATE SCHOOL OF NATURAL AND APPLIED SCIENCES

WIRELESS MULTICAST STREAMING

by
Emin GENÇPINAR

January, 2008
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WIRELESS MULTICAST STREAMING

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by
Emin GENÇPINAR

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We have read the thesis entitled "**WIRELESS MULTICAST STREAMING**" completed by **EMİN GENÇPINAR** under supervision of **ASSIST. PROF. DR. ADİL ALPKOÇAK** and we certify that in our opinion it is fully adequate, in scope and in quality, as a thesis for the degree of Master of Science.

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WIRELESS MULTICAST STREAMING

ABSTRACT

This thesis covers performance analysis of Multimedia Broadcast Multicast System (MBMS) streaming delivery method on emulated UMTS environment considering Reed Solomon Forward Error Correction (FEC) algorithm, tune-in delay, rebuffering effect, MPEG-7, Electronic Service Guide (ESG).

This thesis introduces MPEG-7 based ESG for mobile TV that provides a multimedia query for MBMS services and sessions, retrieves a tree view of available services and a categorized view according to the genre grouping criteria. The prototype covers OMA BCAST ESG fragments and extends content fragment of ESG by MPEG-7. The proposed ESG prototype has been developed using Visual Studio .Net 2005 Smartphone Emulator.

The thesis also covers research on YouTube codecs, interfaces and whether YouTube and MBMS would work together. Another thesis research covers comparison and testing of Xenon Streamer and Darwin Streaming Server (DSS).

Keywords: ESG, FEC, MBMS, MPEG-7, mobile TV, multimedia query, performance analysis, rebuffering effect, tune-in delay, UMTS emulation, wireless streaming.

KABLOSUZ ÇOKLU İLETİŞİM YOLUYLA DURAKSIZ AKIM

ÖZ

Bu tez emülasyonu yapılan UMTS ortamı üzerinde Çoklu Medya Tüm Kamuya Çoklu İletişim ve Gruba Çoklu İletişim Sistemi (MBMS) duraksız akım dağıtım metodunun Reed Solomon FEC algoritmasıyla elde edilen performans analizini, gözlenen tune-in gecikmesini, rebuffering etkisi gecikmesini, MPEG-7 'yi ve elektronik hizmet rehberini içermektedir.

Bu tez mobil TV 'ler için tasarlanan MBMS hizmetleri ve oturumlarının çoklu medya sorgulamasını içeren, sunulan medyanın türüne göre hizmetleri sınıflandıran ve ağaç görünümünde listeleyen MPEG-7 tabanlı elektronik hizmet rehberini tanıtır. Hazırlanan prototip OMA BCAST ESG parçalarını içerir ve ESG 'nin içerik tanımını sağlayan parçasını MPEG-7 kullanarak genişletir. Önerilen ESG prototipi Visual Studio .Net 2005 Smartphone Emulatorü üstünde geliştirilmiştir.

Tezde ayrıca YouTube codec 'leri, arayüzleri ve YouTube ile MBMS 'nin beraber çalışıp çalışmayacağı üzerinde yapılan araştırma verilmektedir. Xenon Streamer ve Darwin Streaming Server (DSS) 'ın karşılaştırması ve bu sunucular üstündeki karşılaştırma testleri de sunulmaktadır.

Anahtar Sözcükler: Çoklu medya sorgulama, ESG, FEC, kablosuz duraksız akım, MBMS, MPEG-7, mobil TV, performans analizi, rebuffering etkisi, tune-in gecikmesi.

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CHAPTER ONE

INTRODUCTION

Wireless multimedia distribution has become very popular with the availability of new technologies. People want to capture photos and videos, store and share them easily with their digital mobile phones. Pervasive and ubiquitous computing affected multimedia devices to become smarter, smaller and mobile. Today both streaming and downloading services are offered over point-to-point wireless connections. Large scale distribution of media makes this point-to-point approach inefficient especially for wireless networks. The recent development in multimedia applications with a parallel progress in wireless transport technologies has brought real-time and non-real time multimedia distribution in the form of multicasting and broadcasting. Such multimedia distribution mechanisms include streaming and downloading services for on-demand video, mobile TV, short clips for news, football results, software updates and more.

Several technologies that provide broadcasting and multicasting for wireless networks are 3GPP MBMS (Multimedia Broadcast and Multicast System) (3GPP, 2007) (OMA BCAST, 2006), 3GPP2 BCMCS (Broadcast and Multicast System), DVB-H (Digital Video Broadcast for Handhelds) that takes an advantage of high bearer rate network, and MediaFLO among others (Mobile TV UMTSF/GSMA Joint Work Group, 2006). MBMS is a multicast and broadcast distribution technology for Third Generation (3G) Universal Mobile Telecommunications System (UMTS) wireless networks (3GPP, 2006). MBMS enables point-to-multipoint transmission of multimedia data by using the existing wireless networks with MBMS specific transmission procedures. MBMS can work with low bearer rates, so consumes less resource allocation and bandwidth.

The main reasons behind the increasing interest in the broadcast and multicast service distribution is the independency from the number of users as well as the resource savings unlike unicast (i.e. point-to-point) packet switched streaming (PSS) for the same transmission power range. In other words in MBMS the number of concurrent users may be limited only by the base station maximum capacity.

There are two delivery modes defined within MBMS, the download and the streaming delivery mode. Download delivery mode is for delivering discrete objects like files where transmission reliability is important. Streaming delivery mode is for delivery of continuous media. For streaming, the video quality is important, but reliability is not compulsory. For either mode there is a need for service discovery or announcement, or a service guide mechanism (OMA BCAST, 2006). A detailed view of MBMS system is given in (3GPP, 2006).

Content description is a critical component of service offering. Service Guide (SG)'s are used by content providers to describe the services, how to access those services and content they make available for offering subscription or purchase of an item over broadcast or interaction channel. SG's are user entry points to discover the currently available or scheduled services and content. SG needs to be refreshed periodically to make functionality consistent (OMA BCAST, 2006). ESG means that SG is electronically available on digital form. EPG (Electronic Program Guide) on the other hand is on screen program guide first used as cable TV guide. Today it is also used by satellite TV applications. Twenty four hour program guides are carried by TV guide channels. EPG data is used with a graphical user interface to view some content descriptions like program titles, start and end times, categorization of services according to channel or genre grouping. ESG covers EPG, because it describes how to access the services, how to purchase or subscribe to items.

The ISO/IEC Motion Picture Group (MPEG) issued in 2002 a standard, called MPEG-7, which enables the content description of multimedia data in XML. The standard supports applications to exchange, identify, and filter multimedia contents based on MPEG-7 descriptions.

Streaming is about real time performance of the system. There are some common streaming delivery Quality of Service (QoS) challenges like the choice of efficient codecs, bandwidth utilization, network latency, packet loss and more.

The network conditions like packet loss are not same for each client within the same coverage area. However, all the clients in a particular multicast session play the same multimedia stream synchronously because of multicast nature of MBMS radio bearers. In MBMS, Forward Error Correction (FEC) can be used to increase reliability hence quality of streaming service.

Both MBMS streaming and download delivery methods can use FEC during the transmission. For high quality streaming and for an error free reliable download delivery, there are several FEC schemes. Important factors of these FEC mechanisms are their encoding/decoding efficiency and their time complexity. Algorithm time complexity particularly is important for the limited processors of handsets. All the FEC schemes have some common points. Encoding symbols which are repair and source symbols respectively are generated during encoding process. A block of “k” source symbols constitutes a source block. Decoding algorithms allow the recovery of the “k” source symbols from any set of the “k” received symbols.

Raptor FEC is the only recommended FEC method for MBMS that is selected by Third Generation Partnership Project (3GPP) community (3GPP, 2006) and it provides linear encode/decode time (Luby, 2005), (Digital Fountain, 2006). Raptor is a fountain code, i.e. as many symbols as needed can be created unlike Reed Solomon (RS) which is a well known and widely used FEC algorithm of which block size includes at most 255 symbols. Raptor decoding time is independent of packet loss patterns. However, Reed Solomon decoding time is loss dependent. Raptor is based on irregular low-density parity-check code (LDPC), since the LDPC codes allow data transmission close to the theoretical maximum (Luby, 2005). Both Raptor and Reed Solomon codes are systematic, so the original source symbols are sent intact from sender to receiver.

1.1 Aim of Thesis

The thesis covers several studies of which one is MPEG-7 based service guide for mobile TV. This study proposes an MPEG-7 based ESG within MBMS that covers OMA BCAST ESG fragments, but extends content fragment of ESG by MPEG-7 to apply content semantics to the services. To demonstrate the usefulness of the new broadcasting environment, a test application in MBMS platform over the UMTS network emulation has been implemented with experiments with two cases including real time content filtering and content based retrieval. Another study is MBMS streaming performance analysis. The study is based on the emulation of an MBMS network to observe the performance of multicast streaming services. Observed tune-in delay and rebuffering effect delay are analyzed to see how streaming is impacted for MBMS users with and without FEC when different network bitrate and loss cases are considered. Vidiator's MBMS streaming system and NetEm network emulator were used to generate Global System for Mobile Communications (GSM) Enhanced Data Rates for Global Evaluation (EDGE) Radio Access Network (GERAN) and UMTS Terrestrial Radio Access Network (UTRAN) conditions. Also in this thesis study, YouTube announcements, codecs, interface and to investigate how to make compatible with MBMS are covered. Another research subject covered comparison and testing Darwin Streaming Server and Xenon Streamer.

1.2 Thesis Organization

This thesis is organized as following; second chapter gives the definition of streaming. Following chapter three describes mobile TV. Chapter four identifies well known competing cellular networks. Following chapter five gives a look to competing wireless broadcast third generation technologies. In chapter six, forward error correction (FEC) is identified and well known FEC schemes are described. Simulation and emulation is explained with a chapter seven, which is important to emulate wired and wireless platforms to enable performance tests. The following chapter eight is related to chapter seven that describes bridging and routing in Linux environment. A content description issue of multimedia world is described in chapter nine, which gives a look to MPEG-7 and OMA BCAST ESG (electronic service guide). YouTube is detailly described in chapter ten, as well as looking for whether YouTube may serve through MBMS or not. Comparison tables about Darwin Streaming Server and Xenon Streamer are given in chapter eleven. Chapter twelve is about related works over these thesis subjects. The following chapter thirteen is about thesis experiments and the last chapter fourteen concludes and gives a look to future work on these research studies.

CHAPTER TWO

STREAMING

Streaming is a distribution of continuous objects like video and audio in a real time. It differs than downloading where distribution of objects like files is discrete and transmission reality is an important and an obligatory factor. As opposite, the streaming relates to the quality of transmission which already there exists some solutions to overwhelm transmission quality issue, such as FEC, Dynamic Bandwidth Adaptation (DBA) or else. Streamed video may not be watched once again, since it is not downloaded to the user disk at all. Each time a user starts streaming and each time the video file units will be streamed.

Wired and wireless way of streaming differs than each others. Wired transmission is still quite faster than wireless communications. Or a few new technologies could be matched with the wired DSL connection speeds. What we care about are cellular networks and expected maximum bearer rate (i.e. network bandwidth) could be 300 kbit/s or a bit more when UTRAN is taken into account.

2.1 Wired and Wireless Streaming

Wired communication makes it easy to download and share video because of high bearer rates of wired medium. Today popular video sharing sites YouTube, SoapBox and others already success this through wired medium. The huge amount of video files in YouTube servers and on demand video progressive downloading mechanism makes YouTube to success that much. However, when an underlined video is tried to be downloaded by everyone, some one will wait for a long time to download a video to watch. YouTube‘s way is to serve users with a progressive downloading solution. That is the reason why again the same video can be watched in YouTube, actually watching the internal copy of the video on a user disk. People day by day enjoy using portable, mobile medium. That brings wireless technologies once again to the top of headlines. Maybe unicast way of progressively downloading (like YouTube) or unicast streaming (either on demand or real time) may be more viable with wired transmissions due to the huge amount of video files and at high Digital Subscriber Line (DSL) bearer rates. Even so, broadcasting or multicasting to a particular small (local) or wide (country) area is easier to implement with wireless networks. The reason is that the transmitted signals over the air transmit all directions that every receiver will easily capture. However broadcasting in wired networks can success by the usage of relays. Relay is a medium that receives unicast connections from servers and multiples connections to each connected hosts that belong to the underlying broadcast group. When mobile TV on cellular networks starts to be common, then we will see that base stations are standing the role of the relay.

CHAPTER THREE

MOBILE TV

3.1 Mobile TV

Mobile TV is a compound of TV services to subscribers via telecommunication networks adapted to the mobile medium. Mobile TV enables interactive, personalized TV services; i.e. having TV service reminder, offering parental control and user friendly features. Program guides and contents are offered by providers to enable users to select which channel to watch.

Mobile TV services are already delivered over point-to-point (PTP) connections. As Figure 3.1 implies, content server delivers content to each recipient via a separate PTP connection.

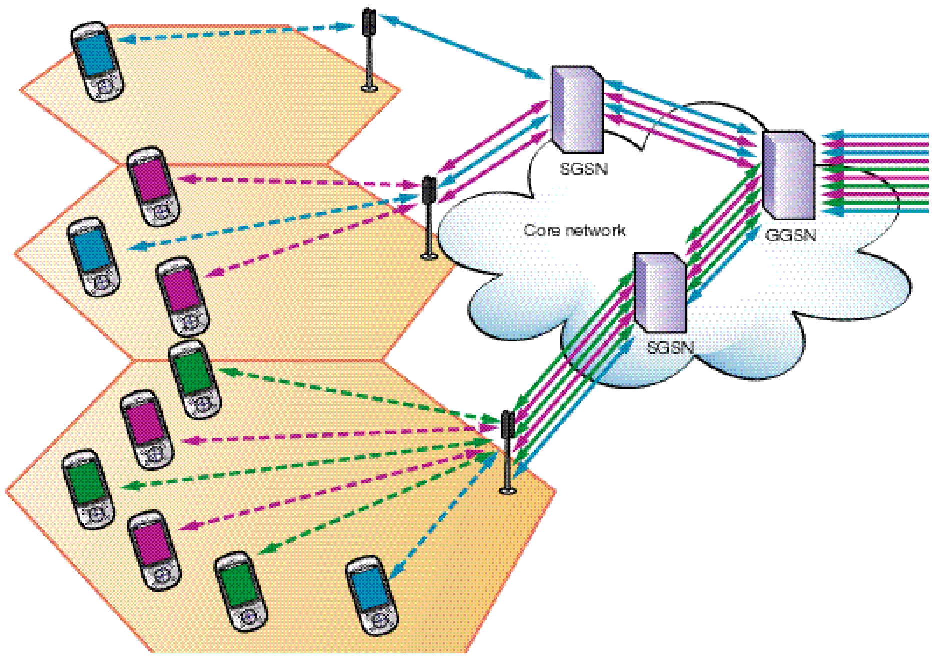


Figure 3.1 The point-to-point (PTP) multimedia delivery via cellular network (Bakhuizen, M., Horn, U., 2005).

When there exists a huge number of load at the same time to download or stream the same media in a particular area, then this PTP solution does not operate well, since that requires more resource allocation at server that also causes huge amount of traffic load at the core network.

Broadcast or multicast is a way of point-to-multipoint (PTM) delivery of multimedia simultaneously from a single source to the multiple destinations. When the same scenario occurs by using broadcast delivery, the server will distribute only one stream per channel in the core and radio network. At the scenario given with Figure 3.2 corresponding to Figure 3.1, the content server will handle three simultaneous streams for all clients.

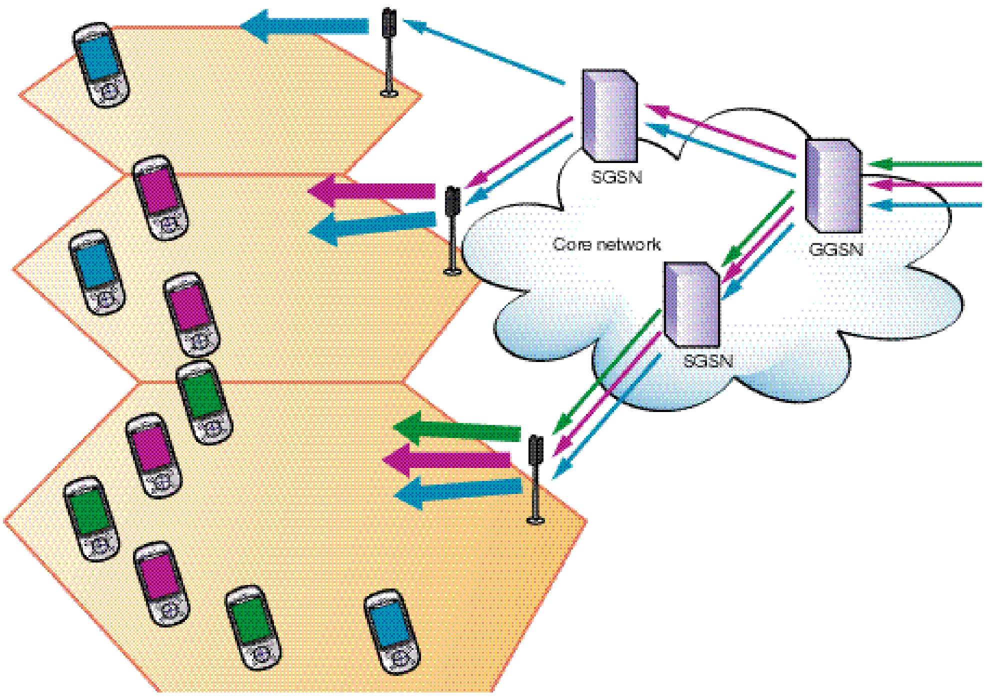


Figure 3.2 The point-to-multipoint (PTM) multimedia delivery via cellular network (Bakhuizen, M., Horn, U., 2005).

Mobile TV delivery is different than traditional way of TV broadcasting (via satellite, terrestrial and cable networks). Mobile TV services are on-demand and upon live video streaming. Mobile TV can be delivered via a two way cellular network or one way dedicated network. Using existing 3G network is the easiest and the fastest way for mobile TV. These are some of the mobile TV standards;

- GPRS; General Packet Radio Service is a packet switched network that enables multiple users to share the same transmission channel. GPRS is used for services as Wireless Application Protocol (WAP). GPRS is the most ubiquitous wireless data service available almost with every GSM networks. GPRS is based on Internet Protocols with throughput rate (about 40 kbit/s) which is a similar access speed to a dial-up modem. GPRS presents customers color Internet browsing, e-mail service, visual communications as video streaming, MMS (Multimedia Messaging Service) and location based services (Wikipedia, 2007). GPRS is used as an underlying technology for SMS (Short Messaging Service), MMS, WAP, internet related services like web browsing and email. GPRS consists of two network structures; GGSN (Gateway GPRS Support Node, a gateway before SGSN) and SGSN (Serving GPRS Support Node) which sends data packets to RNC (Radio Network Controller) or BSC (Base Station Controller). GPRS is a packet switched rather than circuit switched network which means connection oriented.
- DVB-H; Digital Video Broadcasting – Handheld is the transmission system using ETSI Digital Video Broadcasting standards to provide for carrying multimedia services over digital terrestrial broadcasting networks to handheld terminals (ETSI, 2004). All suitable DVB-H spectrums are already being used by analog or digital TV services. Generally these spectrums are assigned to TV services only, which means these cannot be used by other multimedia distribution intents (Bakhuizen, M., Horn, U., 2005).
- CMMB; China Mobile Multimedia Broadcasting which is announced in October 2006 is a mobile TV and multimedia standard developed and specified by the State Administration of Radio, Film and TV (SARFT) which is based on the satellite and terrestrial interactive multiservice infrastructure (STiMi). The CMMB system considers both satellite transmission and additional ground transponders, which are deployed in urban areas and tunnels where satellite signals are weak (Interfax China, 2007).
- MediaFLO; Qualcomm’s new proprietary solution to broadcast data to mobile handsets. The FLO suffix stands for “Forward Link Only” which means that the data transmission path is a one way dedicated from the tower to the

device. The MediaFLO system transmits data on different frequencies than cellular networks (Wikipedia 2007).

- ISDB-T; Integrated Service Digital Broadcasting is the digital TV and radio format that Japanese organization ARIB has created to allow radio and TV stations to convert to digital. ISDB-T (ISDB-Terrestrial) is a key component of ISDB system. Only two countries have adopted ISDB-T. ISDB-T integrates multiple types of digital content as High Definition TV (HDTV), Standard Definition TV (SDTV), sound, graphics, and text. 1seg is the standard in Japan using ISDB (Wikipedia 2007).
- T-DMB; Terrestrial Digital Multimedia Broadcast which is based on the Eureka 147 Digital Audio Broadcasting (DAB) standard is developed in South Korea and can operate via satellite (S-DMB) or T-DMB transmission. T-DMB is an ETSI standard and uses MPEG-4 Part 10 (H.264) for the video and MPEG-4 Part 3 BSAC or HE-AAC V2 for the audio. DMB uses OFDM-DQPSK modulation to deal with channel effects as fading and shadowing. T-DMB works flawlessly in 80 km/h moving vehicle while both TV and radio work fine (Wikipedia 2007).
- 3GPP2 BCMCS; Third Generation Partnership Project 2 Broadcast Multicast Service
- 3GPP MBMS; Third Generation Partnership Project Multimedia Broadcast Multicast Service

CHAPTER FOUR

CELLULAR NETWORKS

4.1 Cellular Networks

The first wireless generation (1G) technology was the invention of the analog cell phones (i.e. mobile phone in US). With second generation (2G) cellular network technology, the digital cellphones started to be used. 2G plus faster data services like GPRS now constitute the 2.5 G of mobile technologies.

- GSM; Global System for Mobile Communications is a digital circuit switched cellular network for transmitting mobile voice and data services. GSM uses narrowband TDMA allowing 124 channels (each channel 200 kHz) and eight users per channel (25 kHz time slots for each user) (What is GSM?, 2007). GSM operates at the 900 MHz, 1800 or 1900 MHz frequency band. GSM is the 2G mobile standard and still widely used in European countries and others such that GSM subscribers can continue to communicate in other countries using global roaming facilities of GSM even changing operators.
- UMTS; Universal Mobile Telecommunications System is a third generation (3G) cell network technology. It also is called as a 3GSM which combines 3G technology and GSM standard to the future. UMTS is standardized by 3GPP. UMTS network consists of three parts; Core Network (CN), UMTS Terrestrial Radio Access Network (UTRAN) and User Equipment (UE). UMTS CN architecture is based on GSM network as well as GPRS (Overview of The UMTS-Draft 2002, 2007). Figure 4.1 depicts the example of UMTS 3G network layout. Node-B stands for Base Station (BS).

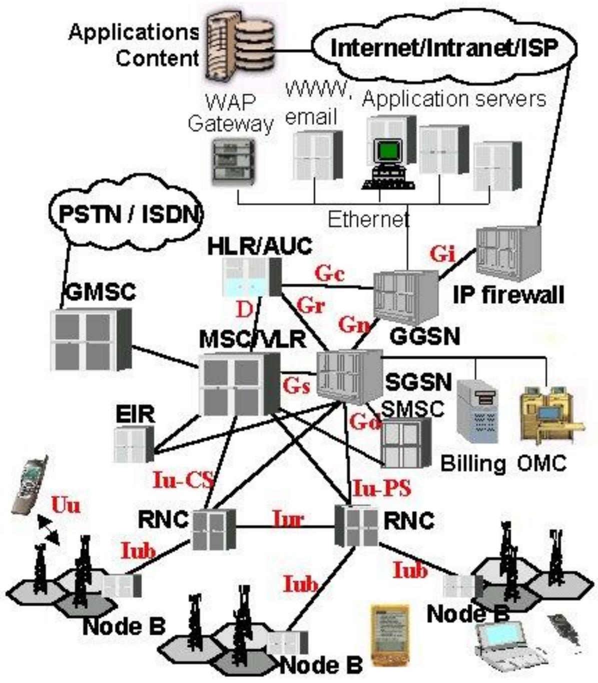


Figure 4.1 UMTS 3G network (Overview of The UMTS-Draft 2002, 2007).

UMTS network provides four different QoS (Quality of Service) classes.

- Conversation class which provides voice, video telephony and video gaming.
- Streaming class which provides multimedia, video on demand and webcast.
- Interactive class which provides web browsing, network gaming, database access.
- Background class which provides email, SMS and multimedia downloading.

UMTS CN provides both circuit switched and packet switched connections.

- GPRS; General Packet Radio Service
- TDMA; Time Division Multiple Access is a method of cellular communications that divides time into slices and assigns whole frequency to one medium at this time slice for transmission. That enables every transmission to continue during own time slice and several users to share the same frequency channel at different time slots, as also depicted in Figure 4.2.
- CDMA; Code Division Multiple Access originally known as IS-95 is spread spectrum cellular technology that competes with GSM and provides shared particular code to a group of users like that people sharing and speaking

different languages. So while many codes are occupying the same channel, only the users associated with the same code can understand each other. Inside several variations of CDMA, original one is also called as CDMAone. Now there exists CDMA2000 and variants like 1X EV, 1XEV-DO, and MC 3X. As depicted in Figure 4.2, CDMA allocates the entire spectrum at all the time to a user and identifies connections by codes (CDMA Overview, 2007).

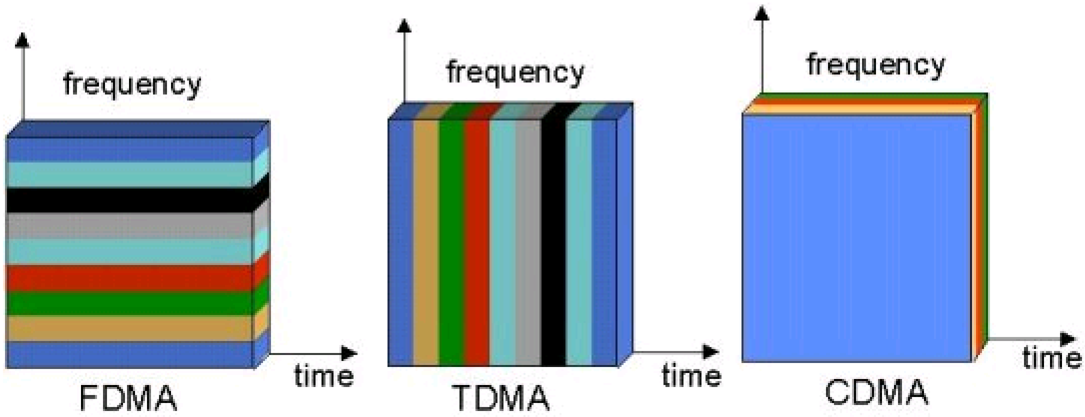


Figure 4.2 Multiple/Medium Access Technologies.

- EV-DO; Evolution Data Optimized/Only is a 3G wireless broadband internet access technology which is standardized by 3GPP2 as part of the CDMA2000 (Wikipedia, 2007). EV-DO was developed by Qualcomm in 1999 to meet over 2Mbps (similar to DSL; Digital Subscriber Line which is a wired technology) as a transmission target speed to meet IMT-2000 requirements. EV-DO USB compatible PC cards at the industry supply EV-DO high speed wireless broadband internet connection to a computer to make it portable like laptop. The competitor technology in US is HSDPA.
- EDGE; Enhanced Data Rates for Global Evaluation or Enhanced GPRS (EGPRS) is a 3G network packet switched technology, which is also intended to be developed as a circuit switched. EDGE does not require any hardware or software changes to be made in GSM core networks, but base stations must be modified to make EDGE compatible. Base Station Subsystem (BSS) must be upgraded. EDGE allows 384 kbps data transmission speed to be achieved when eight timeslots (maximum 48 kbps per slot) are used. EDGE

uses same TDMA frame structure as GSM does and allows existing cell plans intact. The most valuable reference would be (Ericsson White Paper, 2007).

- DECT; Digital Enhanced Cordless Telecommunications uses TDMA to transmit radio signals to phones, but is used with a large number of users in a small area.
- iDEN; Integrated Digital Enhanced Network is a 2G technology that has been developed by Motorola. iDEN is based on TDMA and GSM architecture. iDEN integrates two way radio, alphanumeric messaging, wireless packet data.

CHAPTER FIVE

COMPETING WIRELESS BROADCAST 3G TECHNOLOGIES

5.1 Digital Video Broadcasting-Handheld (DVB-H)

DVB-H is based on DVB-T and totally backward compatible to DVB-T. DVB-H system specifies an efficient way of carrying multimedia data over DVB-T considering physical layer, link layer and service information. Multiprotocol Encapsulated Data Forward Error Correction (MPE-FEC) is an optional to improve carrier-to-noise (C/N) performance to allow receiver to be able to deal with different reception situations. The payload of DVB-H is IP datagram encapsulated into MPE-sections. (Faria, G., Henriksson, J., A., Stare, E., Talmola, P., 2006).

5.1.1 DVB-H Architecture, Protocols and Codec

DVB-H carries payloads (IP datagrams or other high layer protocol datagrams in MPE sections) in an MPEG-2 transport stream (TS) by multiprotocol encapsulation (MPE). With MPE each IP datagram is encapsulated into one MPE section. A stream of MPE sections are put into an elementary stream (ES), which means a stream of MPEG-2 TS packets with a particular program identifier (PID). Figure 5.1 gives the protocol stack for DVB-H. Figure 5.2 depicts the DVB-H system architecture. Adaptations to DVB-T toward DVB-H to address necessary supplement for handheld constraints are;

- Time slicing to reduce power consumption and in order to enable smooth and seamless service handover.
- IP datacasting.
- MPE-FEC.
- 4K carrier mode for network optimization to trade off single frequency network (SFN) cell size and mobility.

MPE-FEC was added to DVB-H system to provide time-interleaving, which is necessary to cope with Doppler Effect, and for error correction. Reed Solomon FEC may be applied after time-interleaving in order to protect data. MPE-FEC and time slicing are implemented at the link layer without affecting DVB-T physical layer.

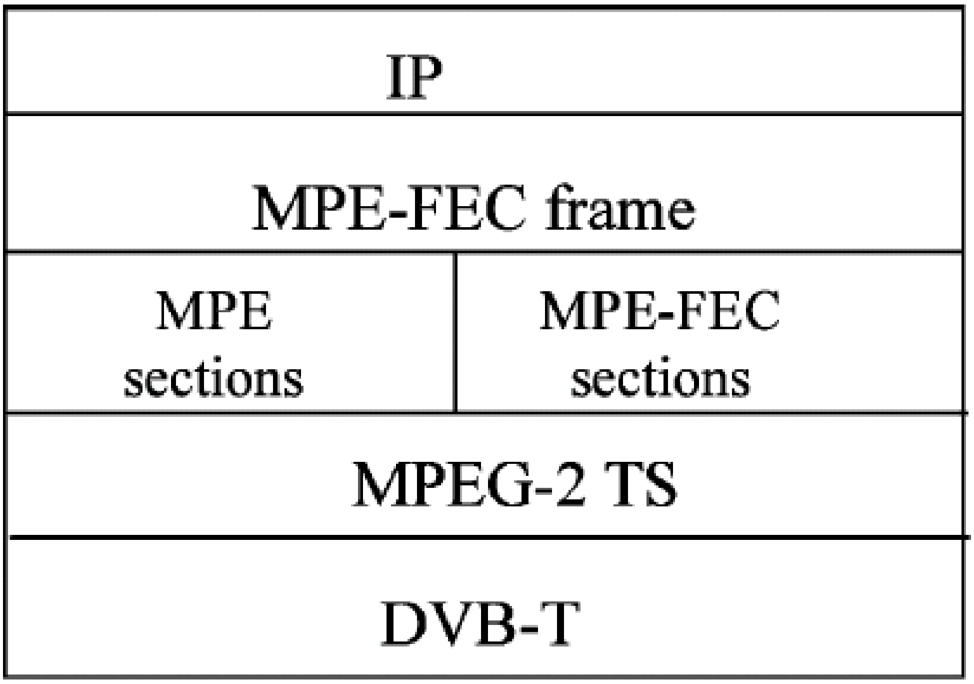


Figure 5.1 Protocol stack of DVB-H (Faria, G., Henriksson, J., A., Stare, E., Talmola, P., 2006).

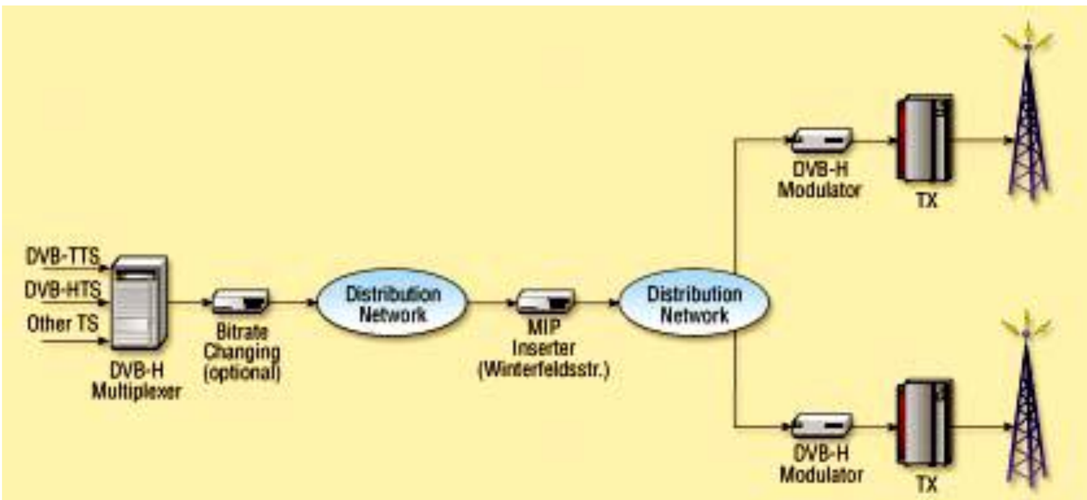


Figure 5.2 DVB-H system architecture.

DVB-H uses HE AAC (High Efficiency Advanced Audio Coding) audio codec.

5.2 Digital Multimedia Broadcast (DMB); T-DMB (Terrestrial DMB)

DMB is a digital radio transmission system in order to deliver mobile multimedia services to mobile handsets. DMB was invented and was first deployed in South Korea. DMB can operate either at satellite (S-DMB) or terrestrial (T-DMB) modes. DMB is based on DAB standard, so deploying T-DMB on existing DAB network will have lower start up cost. DMB resembles competing technology DVB-H. The differences between T-DMB and DVB-H can be summarized by Table 5.1 and Figure 5.3 (Dr. Werling, T., Schepke, C., Dr. Yeun, C. Y., 2005).

Table 5.1 Comparison of T-DMB and DVB-H

T-DMB	DVB-H
• SFN is employed to increase capacity of transmitter. • More efficient usage of frequency resource due to assigning independent frequency range to operators. • Simple receiver structure and robustness to fading. • Coverage signal from DAB and T-DMB are equal. • Very low start up costs for T-DMB with an existing network. • Provide faster channel/program time switch.	• DVB-H could offer higher data rates • Provide more channels services per multiplex. • Complicated receiver structure so prone to fading.

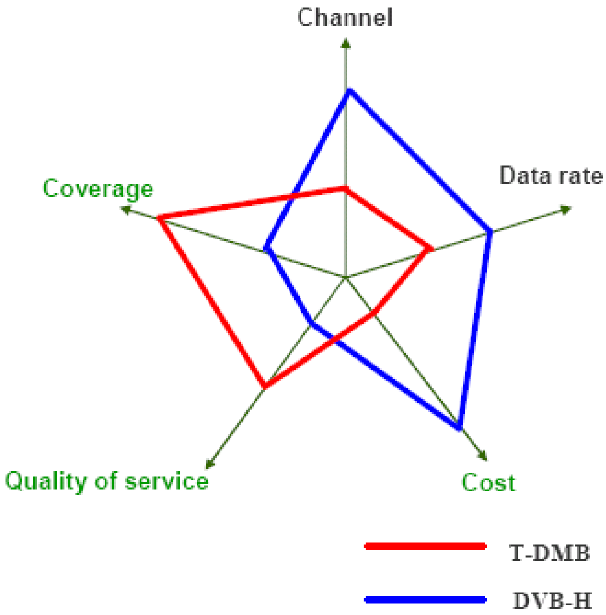


Figure 5.3 T-DMB versus DVB-H.

5.2.1 DMB; T-DMB Architecture, Protocols and Codec

T-DMB uses MPEG-4 Part 10 (H.264) codec for the video and MPEG-4 Part 3 BSAC or HE-AAC V2 codecs for the audio. The audio and video is encapsulated in MPEG-2 TS. T-DMB can carry MPEG-4 BIFS (Binary Format for Scenes) streams. Figure 5.4 gives detailed system architecture of T-DMB (Dr. Werling, T., Schepke, C., Dr. Yeun, C. Y., 2005).

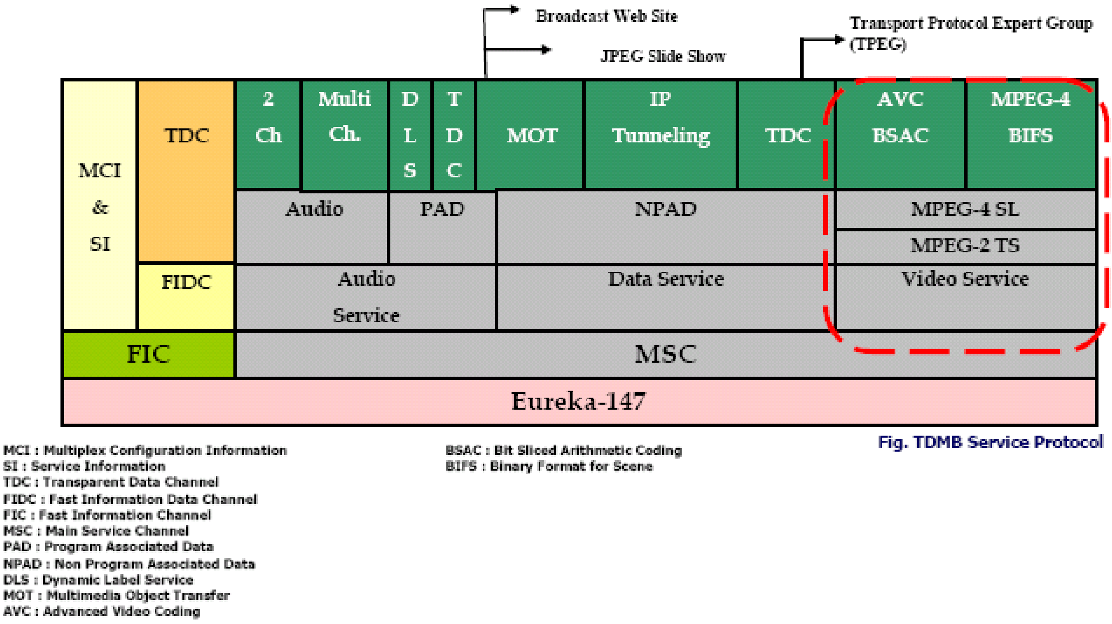


Figure 5.4 T-DMB system architecture.

5.3 MediaFLO

MediaFLO is QualCOMM proprietary solution to 3G networks.

5.3.1 MediaFLO Architecture, Protocols and Codec

Figure 5.5 shows system architecture of MediaFLO (MediaFLO, 2007). The system consists of four components. Network Operation Center which consists of National (NOC) and Local Operation Centers (LOC), FLO transmitters, 3G network and FLO enabled handsets. All the important functionality of FLO network is hidden at Network Operation Center such as billing, content management infrastructure, media distribution, delivery of program and content guide information, access/encryption key distribution. 3G network is used for service subscription, interactive service support to allow mobile devices to communicate with NOC.

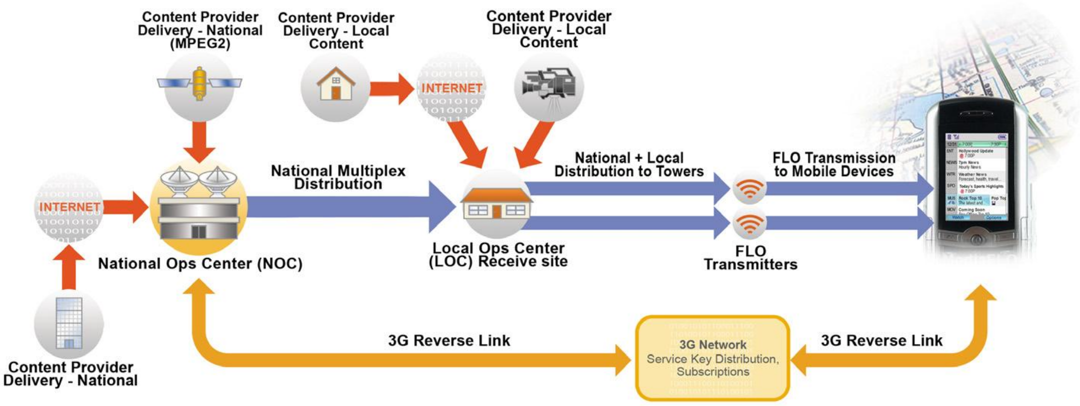


Figure 5.5 MediaFLO system architecture.

MediaFLO uses MPEG-4 AVC and H.264 for video compression. MediaFLO uses AAC+ for audio compression. Figure 5.6 describes a protocol stack that MediaFLO uses.

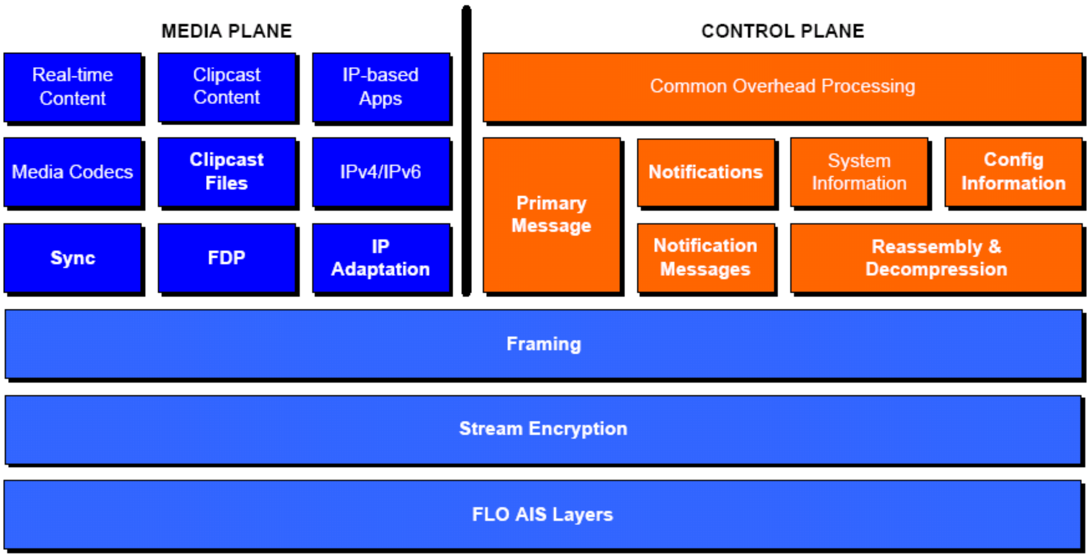


Figure 5.6 MediaFLO protocol stack (Gallouzi, S., 2006).

5.4 Multimedia Broadcast Multicast System (MBMS)

5.4.1 MBMS Architecture, Protocols and Codec

MBMS user services and applications can be delivered either on ptp or MBMS bearers. Figure 5.7 and Figure 5.8 give the protocol stack in MBMS. MBMS broadcast and multicast download and streaming delivery modes are delivered over MBMS bearers.

MBMS speech codecs are AMR narrowband or wideband. MBMS audio codecs are Enhanced aacPlus, Extended AMR-WB. For synthetic audio, the Scalable Polyphony MIDI (SP-MIDI) is supported. For MBMS video, H.264 (AVC) codec is used.

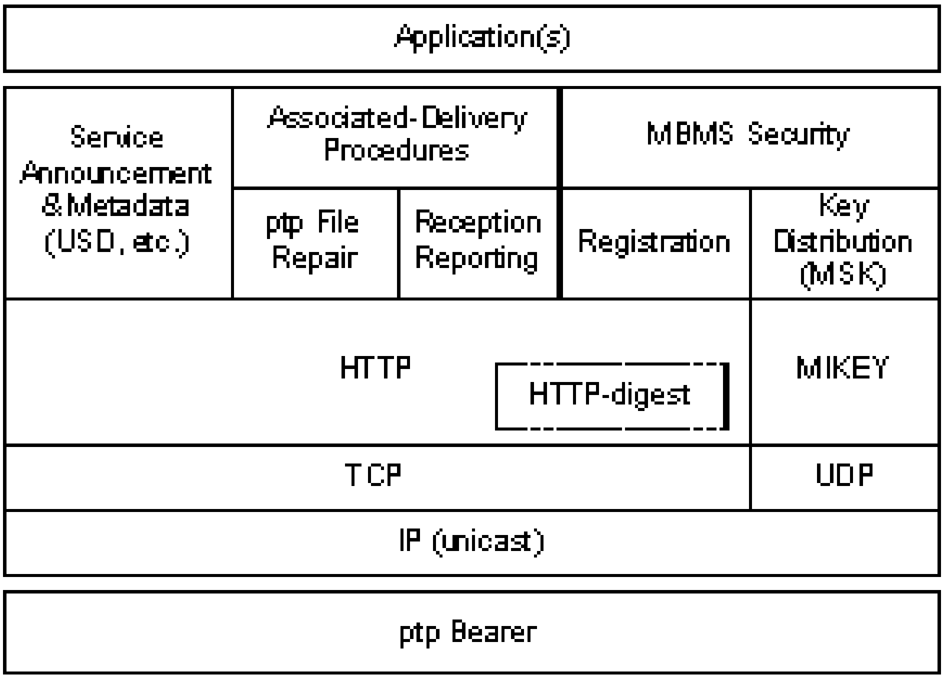


Figure 5.7 MBMS protocol stack on ptp bearers (3GPP, 2006).

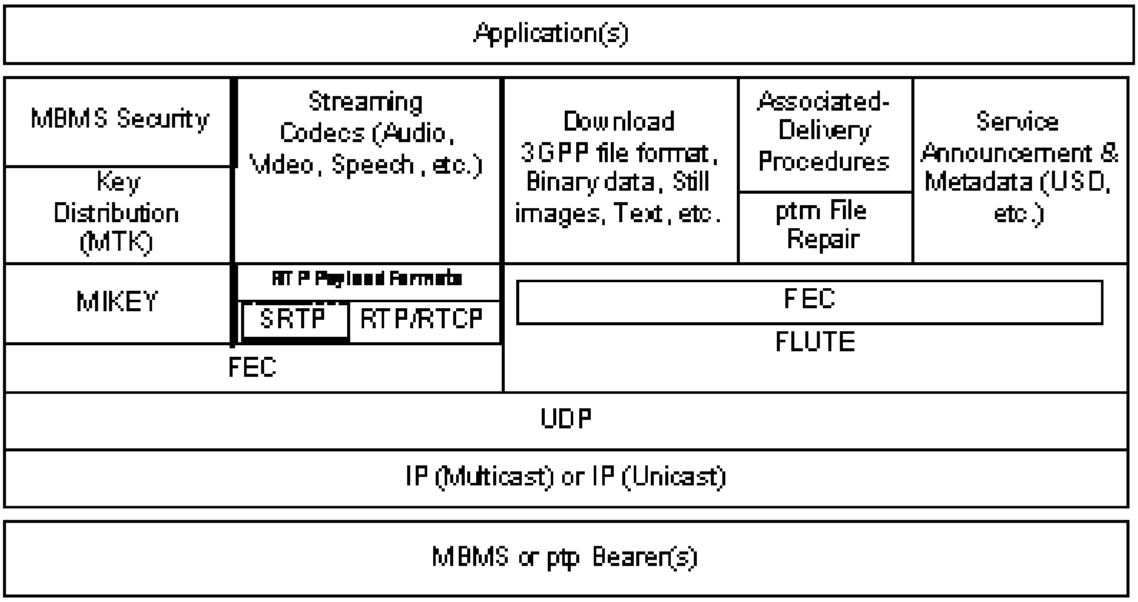


Figure 5.8 MBMS protocol stack on MBMS or ptp bearers (3GPP, 2006).

Figure 5.9 shows MBMS network model. GGSN (Gateway GPRS Serving Node) is a gateway and entry point for MBMS data from BM-SC (Broadcast Multicast Switching Centre). GGSN can be connected with more than one SSGN. BM-SC is used to provide required MBMS functionality.

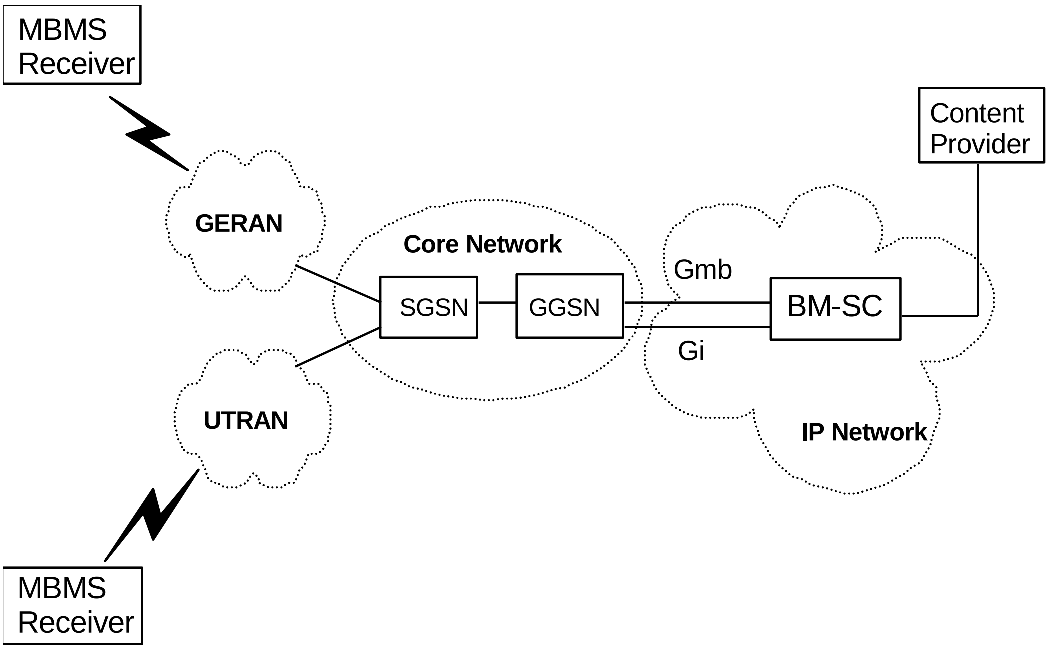


Figure 5.9 MBMS network architecture (3GPP, 2006).

CHAPTER SIX

FORWARD ERROR CORRECTION

6.1 Forward Error Correction

Forward Error Correction is to send a redundant data to the receiver to recover erroneous received or lost data during transmission. Due to the mobility of users, user services should be able to detect and cope with potential data losses. Hence FEC is a critical component of the MBMS system. Figure 6.1 shows the FEC layer in MBMS protocol stack.

For high quality streaming and for an error free reliable download delivery, there are several FEC schemes. Important factors of these FEC mechanisms are their encoding/decoding efficiency and their time complexity. Algorithm time complexity particularly is important for the limited processors of handsets.

A FEC source block is constructed from the source media packets that belong to User Datagram Protocol (UDP) packet flows being relevant to the particular segment of the stream at a time (3GPP, 2006). A source block is a block of “k” source symbols. Source symbols are units of original data that are used during the encoding process. The position of each data from its relevant source packet, a packet that carries only the source symbols, must be indicated within the source block. At last, the repair packets, the packets that carry only the repair symbols, must be determined. The sender side of FEC mechanism will carry both the original packets as source blocks and FEC source packets. After construction of the source block from the original UDP payload together with their flow identity which is identified by destination Internet Protocol (IP) address and UDP port, the FEC encoder generates repair symbols as protection data. The repair symbols are sent at a different channel identified by a different UDP port, this stream is called as FEC repair stream.

The receiver gets the original packets from the media port and buffers them for FEC repair. Buffered packets will later be used to construct the source block. Source FEC Payload IDs of source packets show where to place them in a source block. If some of the packets are lost, but sufficient number of encoding symbols is received, FEC decoder can recover the source block. Otherwise they wait until a timeout occurs where the receivers only deal with the original packets that are available.

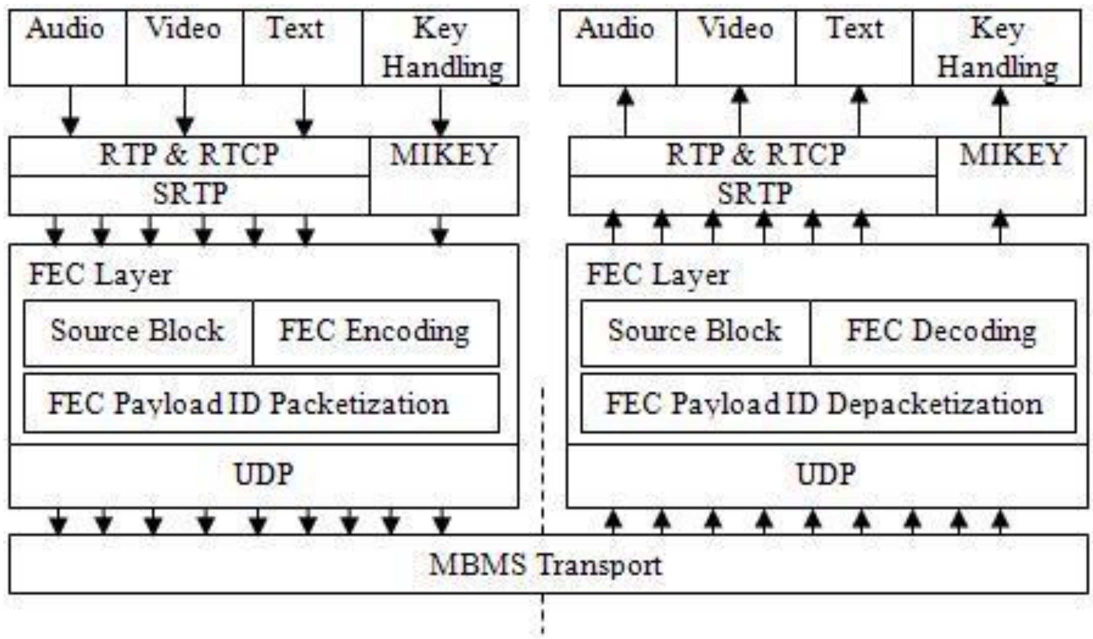


Figure 6.1 MBMS protocol stack for streaming services (3GPP, 2006).

Sufficient number is actually equal to the number of the source symbols in a source block (k). Total amount of FEC repair and source symbols, i.e. encoding symbols, is denoted by “n”.

Within Vidiator MBMS server; Reed Solomon (Lacan, Roca, Peltotalo, Peltotalo, 2007), NULL (Luby, Vicisano, 2004) and Raptor FEC schemes can be applied. Raptor FEC scheme computational complexity is about O(1) time to generate an encoding symbol and O(k) time to decode a message of length “k”.

Wireless networks have limited bandwidth which is an important challenge. FEC improves reliability at the cost of more bandwidth usage. Simply by considering an on-demand streaming service with FEC protection, the server will send more packets than the service not using FEC for the same video and for the same service duration. FEC achieves protection of transmitted data by some redundant data called FEC overhead.

Reed Solomon encoding algorithm computational complexity depends on the current source block length (k) and number of encoding symbols (n) generated for the relevant source block. These parameters are carried by the FEC Object Transmission Information (FEC OTI) to receiver side to execute the decoding algorithm. Reed Solomon is a fully specified FEC scheme of which FEC encoding ID is 129. As explained in (Lacan, Roca, Peltotalo, Peltotalo, 2007), Reed Solomon computational complexities are $O((k / (n - k)) * \log^2 k + \log(k))$ for encoding and $O(\log^2 k)$ for decoding respectively, and each one is about $O(\log^2 k)$ globally.

As mentioned before, while “k” is a number of source symbols, “n” is a number of encoding symbols, “n – k” is the number of repair symbols which are encoding symbols that are not source symbols, then we write,

$$n / k = \textit{FEC redundancy factor.} \qquad (1)$$

$$((n - k) / k) \times 100 = \textit{FEC overhead ratio.} \qquad (2)$$

Then from Equation (1) and Equation (2),

$$(\textit{FEC redundancy factor} - 1) \times 100 = \textit{FEC overhead ratio.} \qquad (3)$$

$$(\textit{FEC overhead ratio} / 100) + 1 = \textit{FEC redundancy factor.} \qquad (4)$$

The number of source symbols is derived as

$$n / \textit{FEC redundancy factor} = k\,. \qquad (5)$$

By converting Equation (1) to Equation (5) we get an inverse relation. According to this inverse relation in Equation (5) between “k” and “FEC redundancy factor”, while “n” is a constant of which value is 255, for increasing FEC redundancy factor, then “k” must be chosen smaller. We know that Reed Solomon encoding and decoding algorithm complexities are $O(\log^2 k)$ globally, so to decrease FEC redundancy factor, then “k” must be chosen higher and RS algorithm computation time is higher. To increase FEC redundancy factor, then “k” must be chosen smaller and RS algorithm computation time is smaller.

Null FEC which has the FEC Encoding ID 0 is a compact no-code fully specified FEC scheme that only produces “n” source symbols. These “n” source symbols at the receiver side generate source blocks without repair symbols. "n - k” is equal to zero, i.e. “k” is equal to “n”, so there is no FEC protection. It is intented to be used for interoperability testing between different implementations of protocol instantiations that use the FEC encoding block (Luby, Vicisano, 2004). With Null FEC, no FEC encoding or decoding is required. Source block includes encoding symbols. Encoding Symbol ID (ESI) is an index of encoding symbol within the source block. When a source block is sent, the Source Block Length (SBL), the Source Block Number (SBN) and the Encoder Symbol ID (ESI) are generated. The FEC Payload ID and the encoding symbol are placed into the packet to send. The FEC Payload ID consists of SBN and ESI. If all the encoding symbols in a source block are received, then the receiver can recover the source block.

Raptor FEC is the only recommended FEC method for MBMS that is selected by Third Generation Partnership Project (3GPP) community (3GPP, 2006) and it provides linear encode/decode time (Luby, 2005), (Digital Fountain, 2006). Raptor is a fountain code, i.e. as many symbols as needed can be created unlike Reed Solomon (RS) which is a well known and widely used FEC algorithm of which block size includes at most 255 symbols. Raptor decoding time is independent of packet loss patterns. However, Reed Solomon decoding time is loss dependent. Raptor is based on irregular low-density parity-check code (LDPC), since the LDPC codes allow data transmission close to the theoretical maximum (Luby, 2005). Both Raptor and Reed Solomon codes are systematic, so the original source symbols are sent intact from sender to receiver.

CHAPTER SEVEN

SIMULATION AND EMULATION

7.1 Simulators

7.1.1 NS-2

NS-2 is the version two of Network Simulator (NS) that covers almost every network types, elements and traffic models, which is one of the most known and widely used network simulator. NS uses OtcL (Object oriented version of TcL) as a command and configuration interface. NS is an object oriented simulator which is written in C++ with an OtcL interpreter as frontend.

7.1.1.1 NSE, An NS Emulator

NSE is an emulator extension of NS. NSE is an emulator generally used by wireless network researchers at sensor network research area, but the emulation conditions are not allowed to be modified as in NIST Net or NetEm. NSE is an emulator feature extension of Network Simulator NS. NS is a Linux simulator which is written in C++, that can be used to extend NS, with Object Oriented Version of Tcl (OTcl) interpreter to execute user's NS scripts. NS may work on Windows through "cygwin" Windows to Linux emulator tool. NS users can use the existing or may define new simulated objects which cover applications, protocols, network elements, types and traffic models. NS Network Animator (NAM) is used to visualize the simulation results.

NS emulation facility is given with two modes; opaque and protocol mode. Opaque mode is useful in evaluating the behavior of real-world implementations when subjected to adverse network conditions that are not protocol specific. Protocol mode can be used for end to end application testing, protocol and conformance testing. The interface between the simulator and live network is provided by a collection of objects including tap agents and network objects.

Tap agents embed live network data into simulated packets and vice versa. Network objects are installed in tap agents and provide an entry point for the sending and receipt of live data. Each tap agent can have at most one associated network object, although more than one tap agent may be instantiated on a single simulator node (NSE, 2006).

7.1.2 OpNet

OpNet is a first commercial network simulator. It acts as a network simulator does. But there does not a script language to create simulation environment. It is easier to manage a simulation environment. There exists a powerful interface for animation, modeling and simulation. Wireless network modeler is also provided.

7.2 Emulators

7.2.1 NIST Net

NIST Net (National Institute of Standards and Technology (NIST) Network Emulation Package) works through a table of emulator entries and works over a Linux machine acting as a router. It is implemented as a Linux kernel module extension. NIST Net handles IP or higher network protocols. NIST Net allows to have controlled and reproducible (as a simulation) experiments with network performance sensitive or adaptive applications and control protocols in a simple laboratory setting. Reproducible environment is said to be relatively quick and easy to assemble. NIST Net is an emulator, which considers both of the simulator and live testing issues inside. Live testing is to test real code in a real environment.

Complex performance scenarios for NIST Net are;

- Tunable packet delay distributions
- Congestion and background loss
- B/W (bandwidth) limitation
- Packet reordering / duplication

NIST Net has an X Window User Interface over Linux. The interface allows the user;

- to select and monitor specific traffic streams passing through the router
- to apply selected performance effects to the IP packets of the stream

NIST Net supports user defined packet handlers to be added to the system. Selected flows can be intercepted by this way.

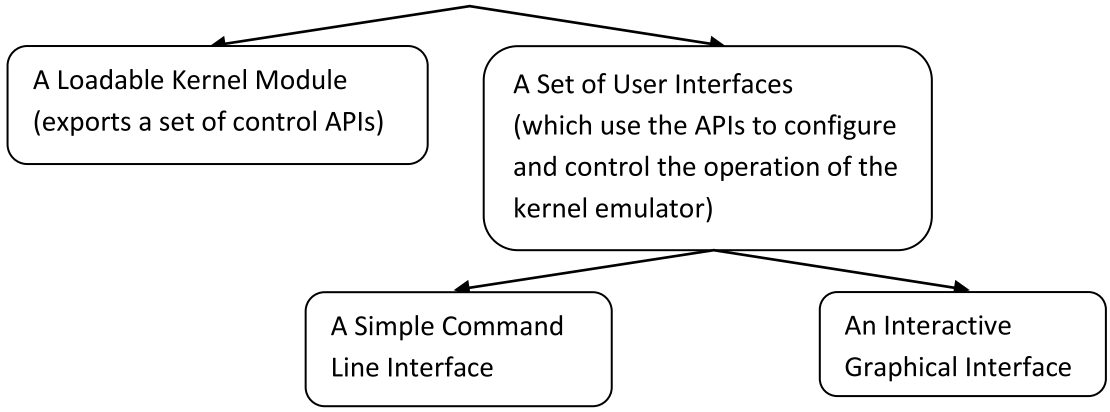
As a kernel design, NIST Net resembles the Sun OS-based Hitbox (Carson, Santay, 2003). NIST Net sometimes is mentioned as a “network-in-a-box”. The important feature of this emulator is that we can selectively apply network effects.

NIST Net works through a table of emulator entries. Emulator entries can be loaded either programmatically or manually. For network effects, NIST Net refers;

- Random distribution for Ordinary (non B/W related) packet delay. By default, “Heavy-tail Distribution” is used.
- Uniform distribution for Ordinary (non congestion related) packet loss and duplication
- DRD (Derivative Random Drop) for Congestion-dependent loss. DRD has a computational simplicity. It drops packets with a probability that increases linearly with the instantaneous queue length after a minimum threshold is reached.

NIST Net uses the DRD congestion control algorithm and allows configuration of the minimum and maximum queue length. DRD is more sensitive to the traffic burst than RED. Both RED & DRD are proactive packet dropping techniques for congestion avoidance. RED drops packets randomly from its buffers if the queue length exceeds a certain threshold. DRD, queue utilization (ratio of service rate to arrival rate) is used to determine when packet drops will occur (Mock, J.H., 2003).

NIST Net Architecture (Carson, Santay, 2003)



- Multiple processes can control the emulator simultaneously
- User interfaces allow controlling and monitoring entries simultaneously

Fast timer is a real time clock for scheduling delayed packets. There is a need to reprogram the clock to interrupt at a sufficiently high rate for fine-grained packet delays.

NIST Net applies Radix Sort to reorder packets. It uses FIFO making the sort stable and eliminating the undesired reordering.

7.2.1 NetEm

NetEm is the most successful and the simplest one. NetEm emulates variable delay, packet loss, duplication and reordering. Bandwidth can be limited by a token bucket filter. It is a protocol independent emulator which works over Ethernet frames. First the Linux bridging must be set as depicted in Table 8.1. Table 7.1 is a sample of how to start emulation over Linux bridging and Table 7.2 may be used to end emulation. NetEm functionality is provided by “tc” command line tool which is a part of iproute2 suite (NetEm, 2006).

Table 7.1 NetEm emulation for bearer rate 128kbit/s and loss 1%.

```
set_bandwidth=128
add_loss=1

echo "Bearer rate (Bandwidth): " $set_bandwidth " kbit/s"
echo "Packet loss / RLC BLER: %" $add_loss

sudo tc qdisc add dev eth1 root handle 1: netem loss $add_loss"%"
sudo tc qdisc add dev eth1 parent 1:1 handle 10: htb default 1
sudo tc class add dev eth1 parent 10: classid 0:1 htb rate
$set_bandwidth"kbit" ceil $set_bandwidth"kbit"
sudo tc qdisc add dev eth0 root handle 1: netem loss $add_loss"%"
sudo tc qdisc add dev eth0 parent 1:1 handle 10: htb default 1
sudo tc class add dev eth0 parent 10: classid 0:1 htb rate
$set_bandwidth"kbit" ceil $set_bandwidth"kbit"
```

Table 7.2 End of NetEm emulation

```
sudo tc qdisc del dev eth1 root
sudo tc qdisc del dev eth0 root
```

7.2.2 Shunra Cloud

Shunra Cloud is a Windows wide area network (WAN) emulator, now with Windows Vista support. It provides a five day free trial. It applies emulation feature before the network packets reach to the final application. Its interface is so easy to use, Figure 7.1. Packet delay, bandwidth limitation and packet loss can be applied.

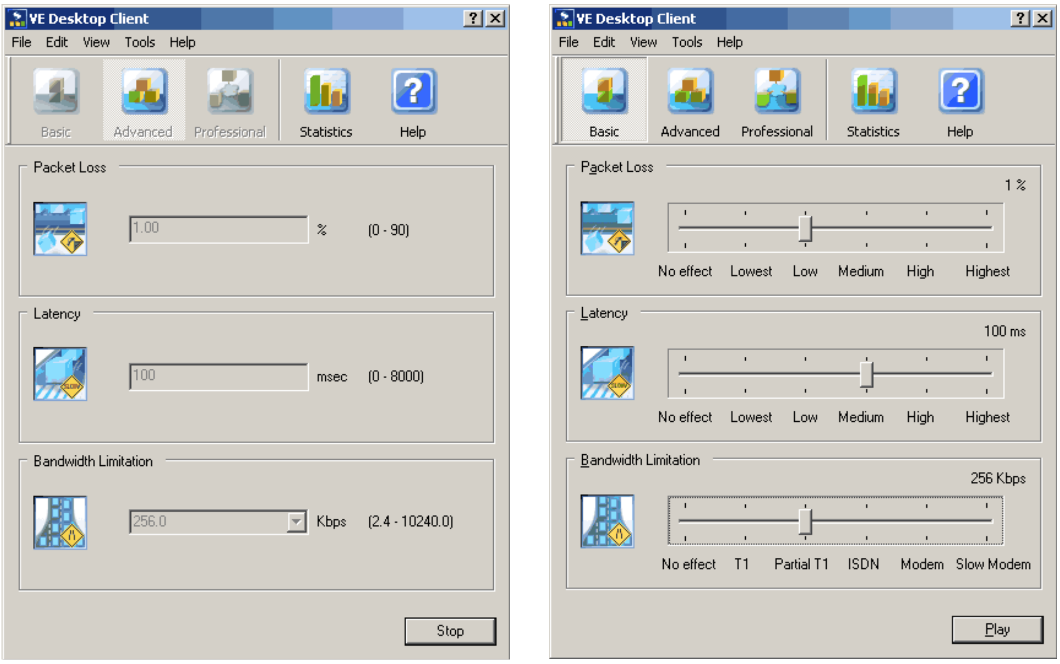


Figure 7.1 Shunra VE Desktop Client user interface (Shunra Ve Desktop download broschure, 2007).

Shunra Cloud would be another solution, if free tool NetEm was not used instead. Shunra Cloud would be installed either server or client hosts. And server application and client player applications would be affected by the emulated network on their listening ports. Already multicast / broadcast packets would reach to client, but while receiving packets, the Shunra emulation tool would loss packets, add delays or would change receiving frequency which means bandwidth limitation. Any bridging would not be required. Shunra would behave like a firewall. As firewall, Shunra Cloud will listen ports, will act your emulation parameters on network and pass emulated network to application.

CHAPTER EIGHT

LINUX BRIDGING AND ROUTING

8.1 Linux Bridging

A bridge is a way to connect two local area network (LAN) segments together in a protocol independent way and is a way of extending a LAN. Packets are forwarded based on Ethernet address, rather than IP address (like a router). Since forwarding is done at data link layer (DLL), all protocols can go transparently through a bridge. Computers do not know whether they are connected to a LAN or a bridged LAN. A bridge can do frame filtering. It does not forward a copy of frame to the other segment if the sender and the receiver are at the same segment. A bridge uses a destination physical address found in the frame headers to forward a frame. Bridges are adaptive which means they can learn a list of computers in each segment. If the LAN supports broadcast or multicast, the bridge must forward a copy of each broadcast and multicast frame to achieve LAN extension operating as a single LAN.

The Linux bridge code implements a subset of the ANSI/IEEE 802.1d standard. Linux bridging code is already grafted into Linux kernel version 2.4 and 2.6 which grabs bridge-utils packages. For implementing linux bridging two network cards must be set on the same computer. If bridge module is set correctly, then brctl command should work. The following setup is used to create bridge between two local networks (Linux bridging, 2007), Table 8.1.

Table 8.1 Setting Linux for bridging

```
ifconfig eth0 0.0.0.0
ifconfig eth1 0.0.0.0
brctl addbr wireless_bridge
brctl addif wireless_bridge eth0
brctl addif wireless_bridge eth1
ifconfig wireless_bridge up
ifconfig wireless_bridge 192.168.2.1 netmask 255.255.255.0
```

8.2 Linux Routing

To use Linux as a router, IP forwarding must be enabled. Router must have at least two different subnets, so Linux host must have at least two NICs.

Table 8.2 Setting Linux for routing

```
sysctl -w net.ipv4.ip_forward=1
```

To enable Linux router to allow multicast and broadcast packets to receive and to release, “mrouted” suite must be installed on Linux (How to set up Linux for Multicast Routing, 2007). In the past years, the Internet was not used for multicasting; many of routers did not support multicasting. Later the solution came with multicast backbone (mbone) suite. That provided a virtual multicast network on Internet. A multicast packet would be encapsulated inside unicast packet and would travel upto the destination network. Then it would again be deencapsulated. So the destination network hosts would receive multicast packet. Actually that would be a multicast tunnel on Internet using unicast encapsulation of multicast packet. What “mrouted” suite will provide is the same, encapsulating multicast packets within unicast packets to send them through unicast routers.

CHAPTER NINE

CONTENT DESCRIPTION

9.1 OMA BCAST ESG

Content description is a critical component of service offering. Service Guide (SG)'s are used by content providers to describe the services, how to access those services and content they make available for offering subscription or purchase of an item over broadcast or interaction channel. SG's are user entry points to discover the currently available or scheduled services and content. SG needs to be refreshed periodically to make functionality consistent (OMA BCAST, 2006). ESG means that SG is electronically available on digital form. EPG (Electronic Program Guide) on the other hand is on screen program guide first used as cable TV guide. Today it is also used by satellite TV applications. Twenty four hour program guides are carried by TV guide channels. EPG data is used with a graphical user interface to view some content descriptions like program titles, start and end times, categorization of services according to channel or genre grouping. ESG covers EPG, because it describes how to access the services, how to purchase or subscribe to items.

OMA BCAST offers ESG architecture for MBMS. OMA BCAST defines Service Guide Delivery (SGD) fragments which are Service, Schedule, Content, and Access respectively. First three fragments obtain core of ESG structure and Access fragment covers session details that also includes some part of the Session Description Procedure (SDP) file. ESG also relates core fragments by Purchase Item. Purchase Item, Data and Channel are provisioning fragments. ESG structure has provisioning, core and access components as shown in Figure 9.1.

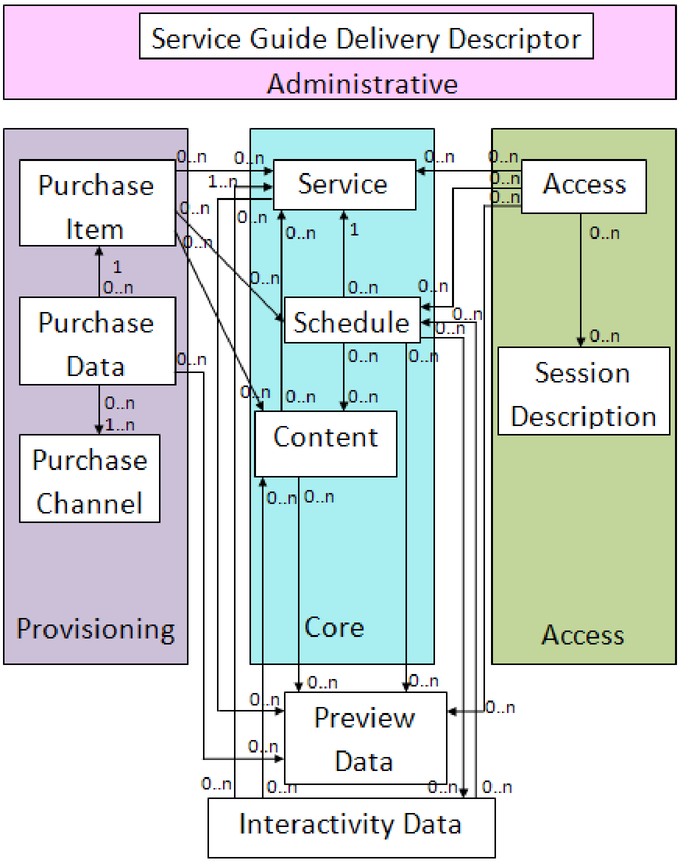


Figure 9.1 OMA ESG data model (OMA BCAST, 2006).

9.2 Extending MPEG-7

MPEG-7 is a multimedia description standard, and endeavor to describe multimedia content in both high level and low level. High level description components of MPEG-7 aim to point out semantic relationships between objects in multimedia content. On the other hand, low level descriptors can be extracted automatically from multimedia content. Besides, some annotation tools for MPEG-7 could be used to annotate multimedia objects manually (i.e. specifying keywords for multimedia content).

In this study, instead of associating MPEG-7 descriptions into ESG, we propose a different approach which extends MPEG-7 to be able to hold some service related information such as scheduling, parental rating etc.

Although annotation tools for MPEG-7 are sufficient to annotate multimedia content, MPEG uses XML schema, and allows MPEG-7 to extend when needed. There are two major ways to extend MPEG-7, which are DType and DSType. Figure 9.2 and Figure 9.3 show DType and DSType in XML schema fragments. Figure 9.4 depicts XSD schema of the Modified Text Annotation Type of MPEG-7, and Figure 9.5 gives a detailed view of the whole Mobile TV architecture, which is extended MPEG-7 Meta Structure, in class diagram format.

```
<complexType name="DType"
abstract="true">
    <complexContent>
        <extension
base="mpeg7:Mpeg7BaseType"/>
    </complexContent>
</complexType>
```

Figure 9.2 XSD schema of the DType.

```
<complexType name="DSType" abstract="true">
    <complexContent>
        <extension
base="mpeg7:Mpeg7BaseType">
            <sequence>
                <element name="Header"
               type="mpeg7:HeaderType"
               minOccurs="0"
          maxOccurs="unbounded"/>
            </sequence>
            <attribute name="id" type="ID"
               use="optional"/>
            <attributeGroup
     ref="mpeg7:timePropertyGrp"/>
            <attributeGroup
     ref="mpeg7:mediaTimePropertyGrp"/>
        </extension>
    </complexContent>
</complexType>
```

Figure 9.3 XSD schema of the DSType.

```
<complexType name="TextAnnotationType">
  <choice minOccurs="1"
     maxOccurs="unbounded">
    <element name="FreeTextAnnotation"
     type="mpeg7:TextualType"/>
    <element name="StructuredAnnotation"
     type="mpeg7:StructuredAnnotationType"/>
    <element name="DependencyStructure"
     type="mpeg7:DependencyStructureType"/>
    <element name="KeywordAnnotation"
     type="mpeg7:KeywordAnnotationType"/>
    <element name="MobileContent"
     type="mpeg7:MobileContentType"/>
  </choice>
  <attribute name="relevance"
     type="mpeg7:zeroToOneType"
     use="optional"/>
  <attribute name="confidence"
     type="mpeg7:zeroToOneType"
     use="optional"/>
  <attribute ref="xml:lang"/>
</complexType>
```

Figure 9.4 XSD schema of the modified text annotation type of MPEG-7.

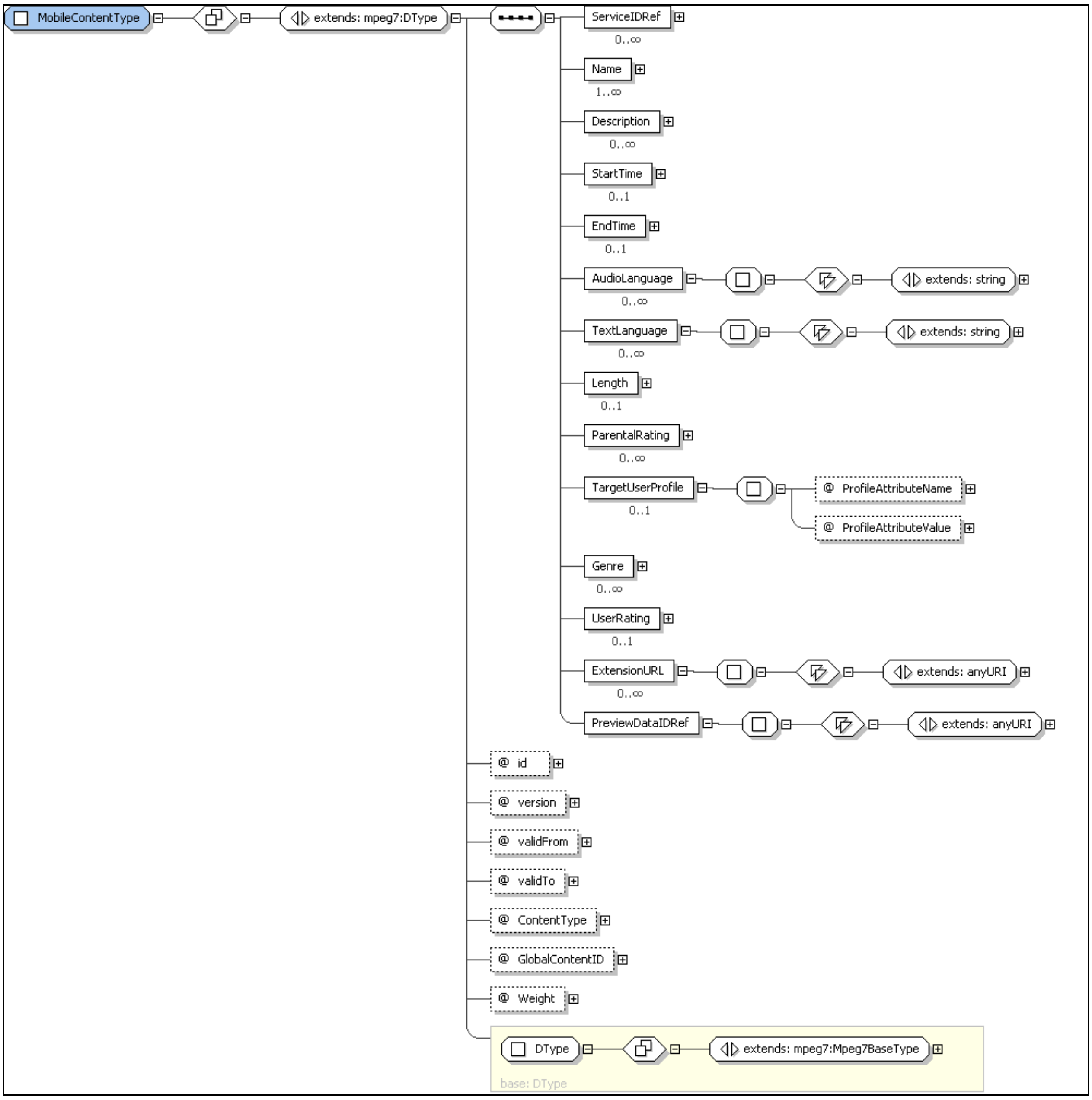


Figure 9.5 Class diagram of Mobile TV extended MPEG-7 meta structure.

CHAPTER TEN

YOUTUBE AND MBMS TOGETHER

10.1 Overview of YouTube

Hurley, Chen, and Karim, whom are YouTube founders, wanted to share some videos from a dinner party with friends in San Francisco in January 2005. Sending the clips around by e-mail was being rejected because they were so big. Posting the videos online got trouble. They needed to develop something easier (Hurley, 2007). You can watch videos on the site without downloading any software or even registering. Google aims to competite local video sharing web sites like DailyMotion in France. It also made an agreement with local television stations like M6 and France Télévisions to broadcast legally the video content. Google also planned to localise in Germany in the future (YouTube Wikipedia, 2007).

To help people share their talent, imaginations and experiences with the world, YouTube users can now become their own Channels. Rather than following the traditional TV model where executives and companies are telling viewers what to watch, YouTube empowers its community to take control and be masters of their own entertainment domain. YouTube allows people to watch what they want, when they want, and receive new videos from their subscriptions to keep up-to-date with their favorite channels (YouTube Launches, 2006).

YouTube enables submitting videos in several common file formats such as ".mpeg" and ".avi". YouTube clone automatically converts them to Flash Video with ".flv" extension and makes them available for online viewing. Flash Video (flv) is a popular video format among large hosting sites due to its wide compatibility. FLV is a file format used to deliver video over the Internet to the Adobe Flash Player version 6, 7, 8, or 9. FLV content may also be embedded within SWF files. Notable users of the FLV format include Google Video, Reuters.com and YouTube. FLV is viewable on most operating systems, via the widely-available Macromedia Flash Player and web browser plug-in, or one of several third-party programs such as Media Player Classic if the ffdshow codec installed, MPlayer, or VLC media player. FLV may also be embeded in web pages. Most FLV files use a variant of H.263 to encode the video. FLV files may contain audio in PCM, ADPCM, or MP3 format. FLV is limited to one video and one audio stream per file (YouTube clone, 2007). Flash Player 7 is used with Sorenson Squeeze codec, Flash Player 8 is used with On2 VP6 codec (About External Progressive Download, 2007). YouTube probably still uses Sorenson Codec because YouTube serves with Flash Player 7 as a minimum requirement.

Unregistered users can watch most videos on the site; registered users have the ability to upload an unlimited number of videos. Related videos are determined by the title and tags. YouTube provides to post video 'responses' and subscribe to content feeds for a particular user or users (YouTube Wikipedia, 2007).

YouTube allows web site developers an API to grab video details by developed API REST interface. Only thing is to support your YouTube developer ID and the requested video ID. The result is an XML document that displays the details of a requested video (check http://www.youtube.com/api2_rest?method=youtube. videos.get_details&dev_id=jRdmPQ4Z0zw&video_id=C7V3WCy1Ozc). Videos may also be uploaded by MMS capable mobile phones. Google uses YouTube and embeds the relevant video from YouTube to the Google results, while the searched keywords are related to the video content on YouTube. YouTube may disallow video to be embedded by the external sites, if the uploader user chooses not to share video with the other sites.

YouTube enables user to add a quick playlist of videos and play all later. The new generations do not care about the difference watching either the internet video or a television.

An important missing point in YouTube of Google, SoapBox of Microsoft like free download sites is that they do not give a chance to search a live streaming video.

You can download the content by external third party softwares as KeepVid tool (KeepVid, 2007) even YouTube does not allow it (YouTube help center, 2007), but you cannot search a television channel. It would be a good opinion if people could watch the television programs by searching site. As they say, people do not care about what the television or YouTube differs, because they can watch the videos from YouTube instead of television. It would be better if the television channels were loaded to YouTube channels. Albeit YouTube is an internet site, the user cannot watch a video while using a computer, e.g. writing a document.

Microsoft SoapBox encourages user to create a free user account, to share personal videos, to search other videos and to add a comment on video as YouTube presents. SoapBox also uses Flash Player as YouTube does. SoapBox can also be embedded to the web pages, and sets always the aspect ratio as 4:3. SoapBox merges video player and searching on one page and disables page refreshing for all page. SoapBox full screen view stands inside the Internet Explorer area while YouTube full screen mode extends on all screen as a foreground application.

People download videos from YouTube (Why does the video…, 2007). The point in Flash Player is the fact that the player starts to download once a user clicks the play button. The video downloading continues even the user pauses the video. The video downloading resembles progressive downloading (About External Progressive Download, 2007). Once the video has been downloaded to the user disk, the video can be watched from the internal copy on local disk even the network is unreachable. As YouTube mentioned (Why does the video…, 2007), at least 500 Kbps broadband connection is required to get the best view.

YouTube system minimum requirements on several platforms (Why can't I hear, 2007) are Macromedia Flash Player 7.0 plug-in, Windows 2000 with latest updates installed, Mac OS X 10.3, Firefox 1.1, Internet Explorer 5.0, Safari 1.0, broadband connection with 500 Kbps.

YouTube successes the way it does because of extensive video number. Total uploaded video number increases, so people rarely watch the same videos at the same time. Too much users search the YouTube videos and watch more than one videos when they are on YouTube, but that does not give an extra load to the YouTube servers generally, but when all the people in one area search the same thing on YouTube at the same time, then the server that holds this video can be inactive due to extreme number of response from users. In such a condition, broadcasting would eliminate such a problem. If a broadcasting feature could be added, then a live multicast chating could be added, then people could be able to chat through YouTube on videos. YouTube can success if the number of videos grows, so people can find a lot of videos to search. There are huge numbers of people in the world that watch YouTube videos, but this is a huge but almost constant number. The big problem is the extreme amount of uploaded videos and their disk wastage on YouTube servers. YouTube receives almost an average huge number of video uploading a day.

YouTube like sites can be used for trailers of videos to recognize and to be able to sell the full video. Such approach would remove user upload feature than YouTube, but it would be better to search and present a video over such kind of tool. As an example, it could be better to watch the trailer video instead of reading an abstract of any proceeding paper through IEEE explorer tool and to comment on the paper.

YouTube makes it possible to watch more than one video at a time, but does not organize these videos on the same page. A free beta version of View Box Player (View Box Player, 2007) divides the screen into two parts to make this possible. View Box Player only focuses on YouTube videos, uses Flash Player, is compatible with Windows NT, XP and Vista. View Box Player has no affiliation with Google and YouTube.

It would be better to see the picture in picture view while watching more than one YouTube videos. It would either be better to watch YouTube in a car. If it could also show live television and radio channels besides videos we tube, YouTube could be thought as a media player of our cars. To see a YouTube enabled television would either be better with people reading and writing comments via television. Windows Media Center is compatible with SoapBox.

LG and YouTube signed that LG will produce YouTube phone (LG to develop YouTube phone, 2007) (LG handsets to provide, 2007). LG phone will provide both accessing and uploading facilities. For either iPod, Mac or PlayStation Portable, the YouTube video formats should be converted for the appropriate codecs and bitrates for each device, as TubeSock does (TubeSock, 2007). YouTube recommends MPEG4 with Divx or Xvid format that has 320x240 resolution using thirty frames per second for video and MP3 for audio, for getting best view on YouTube. YouTube accepts video files in ".wmv, .avi, .mov, .mpg" file formats (YouTube help center, 2007).

CHAPTER ELEVEN

COMPARISON OF DARWIN AND XENON STREAMING SERVERS

11.1 Characteristics and usability comparison of the servers

Servers have some important functional characteristics, as the software features. Usability issue depends on user compliance to the user interface of the server.

The servers must deal with a huge network load, the performance effecting issues, codecs, protocols within different platforms like operating system or hardware differences, interoperability with third party softwares like client softwares, databases or etc. Streaming servers basically must deal with the number of concurrent users, different codecs, and network protocols on different network layers, file formats, error and access logs. There are several streaming servers and we will consider the latest versions of Vidiator's Award Wining Xenon Streamer Server and Apple's Well Known Open Source Darwin Streaming Servers.

Any software can be tested as (Software Testing Types, 2007);

- Compatibility testing: Testing software with different hardware, operating systems and on different platforms.
- Conformance testing: Verifying software conformance to some specified standards, interoperability.
- Functional testing: Evaluating software compliance with specified requirements, testing software functions.
- Performance testing: To test software reliability under maximum peak conditions for different situations.

• Provided or appropriate.
o Not provided or not appropriate.

Table 11.1 is about the comparison of the server characteristics and usability features.

Table 11.1 Darwin streaming server and Vidiator Xenon Streamer characteristics and usabilities

Server Characteristics	Vidiator Xenon Streamer	Apple Darwin Streaming Server
Server Limits	Open Issues are given with the latest release document.	○ Open Issues are unknown, but fixed issues are given with the latest administrator guide. But all issues are discussed by developers and programmers on the mailing lists (Apple mailing lists, 2007).
Supporting Daily, Monthly,Yearly Statistics ?	●	○
Price / License	Available as Trial or Licensed	Open Source Code-Free/Apple Public Source License (APSL, 2007)(Open source Darwin streaming server, 2007)
First public release date	Before 2006	March 16 th, 1999; firstly compiled only on Mac OS X
Stable version and date	XSS Version 4.1.3195, June 18th, 2007	DSS Version 5.5.5, May 10 th, 2007
Server Administrator Panel	○	Server Snapshot; Computer Name, Status and Start Date/Time, Server Current Time, Up Time (Current Time minus Starting Time), DNS Time, Server Version, Server API Version, CPU Load, Current Number of Connections, Current Throughput, Total Bytes Served, Total Connections Served - Server Snapshot does not refresh automatically Connected Users; Page View of Connected Clients, Details of Connected Users as Packet Loss, Video Sent, IP address, bit rate and bytes sent to client. Clients are listed unless they close player window Relay Status; Page View of Connected Relays, Details of Relays as Relay Name, Source Destination , Bit Rate, Bytes Relayed General Settings; Enables to set Media Directory, Max.Number of Connections,

			Max Throughput, Default Authentication Scheme, to change Admin Username / Password, to change Movie Broadcast Password, to change MP3 Broadcast Password Port Settings; To overcome firewall through HTTP port 80, because firewall will not restrict port 80 by default. Relay Settings; To add a new relay, to edit or to delete previous relays. Relay is an another host on the net that the server see as a client. Relay can be used for a wired multicast/broadcast point-to-multipoint transmissions. Relay connections can be saved by user name and password authentication. Log Settings; To set the intervals and maximum amount of disk usage of Error and Access Logs, the choice for logging. Playlists; Tocreate, edit and delete MP3 or Media Playlist, to rate media. Enables to log playlist activities, to play randomly or sequentially media, drag & drop property
			WebAdmin is a Perl-based web server to administer the server (Quick Time streaming server modules programming guide, 2005)
Augmentation for Administrator Panel	Required Functionality	○	●
	Simple Design	○	●
	Ease of Usability	○	●
	Ease of Access	○	Through Web/Localhost, Perl script must be run over streaming server computer, that requires an extra web server to be installed on server
Server Help Tool		Hard copy user documentation	DSS Online Help beside hard copy user/administrator documentation Mailing Lists for developers and users that are actively followed by Apple engineers Frequently Asked Questions (FAQ)

Table 11.2 has some unresolved fields.

❖ Unknown issue or not implemented during thesis concept because of other issues.

Table 11.2 Tests on Darwin streaming server and Vidiator Xenon Streamer

Tests on Servers	**Vidiator Xenon Streamer**	**Apple Darwin Streaming Server**
Compatibility Testing		
Supported Operating Systems	Solaris 9-10, Windows 2003 or later, Linux kernel version 2.4 or later, Red Hat Enterprice 4.0 or later, HP UX 11.0, HP UX 11.i	Mac OS X, Red Hat Linux 7.2 (Red Hat 8 or later recommended), Solaris 9, Windows 2000-2003-NT
Client Computer Minimum Requirements	❖	QuickTime Player 6 and broadband internet connection, ISO compliant MPEG-4 player
Server Computer Minimum Requirements	Ultra SPARC III CPU for Solaris 9 and Ultra SPARC T1 for Solaris 10	128 MB RAM (512 MB and 500 Megahertz for high traffic), 1GB available disk space
Live Broadcasting Requirements	One of these encoders; Xenon Live Encoder (Xenon Streamer 4.1 User Guide, 2007) Nexcaster 2 (a previous version of Xenon Live Encoder) Helix Live Encoder Apple QuickTime Live Encoder Vidiator's Proprietary Smart Live Streaming Protocol (SLSP)	Source equipment for camera, microphone QuickTime Broadcaster or other broadcast software (PowerPC G4 recommended for MPEG-4 broadcasting) - Using only PlayList option can stream media to the clients, and all the clients connect to the same playlist and plays the same frame at a time, but the connection is unicast.
Handset Compatibility	Compatible with any 3GPP R5/R6 or 3GPP2 standard handset	❖
Conformance Testing		
Mobile Delivery Solutions	Fast Channel Switching (FCS) Fast Track Switching (FTS) that is controlled by licensing Optimization for HSDPA streaming Proxy for Helix and Darwin Dynamic Bitrate Adaptation (DBA), conformable with 3GPP Release 5 handset without any special requirement on handset Cache for DBA content if Xenon Streamer is used as a remote server 3GPP Release 5 codec streaming including MPEG-4, H.263, AAC and AMR (Xenon Streamer product information page, 2007) 3GPP Release 6 codec streaming including H.264 and aacPlus 3GPP2 codec streaming including QCELP and EVRC Client adaptation Real-time billing interfaces for flexible service logic and CDR creation Service statistics reporting system Carrier grade integration functionalities like SNMP Personal playlist	❖

Interoperability		Conducted by IMTC PSS-AG (Packet Swicthed Streaming-Activity Group), first tests completed in March 2002.	Conducted by ISMA (Internet Streaming Media Alliance) for MPEG-4 interoperability
Technical Specifications			
Codec	Video	H.264 (AVC) Baseline Profile Level 1b, H.263 Profile 0 (Baseline), Profile 3, level 10, 45 MPEG-4 Visual Simple Profile Level 0~2	Sorenson Video (DSS compatible file formats, 2007) H.263 Motion JPEG A H.261 Animation Cinepak Graphics Motion JPEG B Photo JPEG Video
	Audio	MPEG-4 AAC LC aacPlus Enhanced aacPlus	QDesign Music codec QUALCOMM Pure Voice DVI 4:1 ALaw 2:1 uLaw 2:1 16-bit raw IMA 4:1 MACE 3:1 MACE 6:1
	Speech	AMR-NB, AMR-WB, 13k QCELP, EVRC	❖
Protocols	Signalling	RTSP (RFC 2326), SDP (RFC 2327, RFC 3605, RFC 3556)	RTSP (RFC 2326) (Quick Time streaming server modules programming guide, 2005), SDP, RTSP over HTTP tunnelled
	Transport	RTP (RFC 3550 & 3551) RTP interleaved over RTSP/TCP (RFC2326)	RTP (RFC 1889) RTP over Apples's Reliable UDP RTP over HTTP tunnelled RTP over RTSP/TCP
	Video Payload	MPEG-4 (RFC 3016) H.263 (RFC 2429) H.264 (RFC 3984)	MPEG-4 H.263
	Audio Payload	MPEG-4 AAC, aacPlus and Enhanced aacPlus (RFC 3016 & 3640) MP3 (RFC 3119)	AAC AMR QCELP
	Speech Payload	AMR-NB, AMR-WB (RFC 3267) EVRC (RFC 3558) 13k QCELP (RFC 2658)	❖
File Format		3GPP (.3gp) , 3GPP2 (.3g2), ISMA (.mp4), KWISF (.k3g)	3GPP (.3gp) , 3GPP2 (.3g2), ISMA (.mp4)
Functional Testing			
24/7 Availability		❖	❖

User Interface Comparison	Main View	Console view Eight options to use server - Registering/Deregistering service as Windows NT service - Starting/Stopping service on background - Running service as standalone - Version Information - Setting/Displaying service name Gives the details of platform as operating system, CPU number Loads server’s initialization (ini) file and Loads modules Listens the connections from server RTSP-TCP and HTTP-TCP 8088 ports Lists the connections from the clients by time and destination IP address When the client is connected, the maximum bandwidth is given			Console view No options except than starting service option Loads modules Listens the connections from server ports Does not list the connections from the clients		
Server Installation Convenience		●			●		
		Direct Installation, No Configuration			Requires user name and password, the latest Perl version		
Server Logs		Standard of W3C: Common Log Format (CLF) used with Access Log Server Logs: Error Log, Access Log, Transfer Log (RTSP, RTCP Data, DBA Data Log), Boot Log, SOAP Log			Server Logs: Error Log, Access Log (can be analyzed using third party softwares (Sawmill universal log file analyzer, 2007))		
Database Management		PhoneDB, a database for phone information that does not require any third party database software			❖		
Network Security		User is allowed to provide secure access control to deny or to permit client connections through specific IP addresses. Acl.ini file can be organized according to the specified request (Xenon Streamer 4.1 user guide, 2007).			❖		
Performance Testing							
Performance and Limit Values		Theoretical Maximum	Maximum	Observations	Theoretical Maximum	Maximum	Observations
Number of Concurrent Users		❖	❖	❖	❖	❖	Number of concurrent clients can be watched from http://server_ip:554/modules/admin/server/qtssRTPSvrCurConn - Page does not refresh itself - Page is forbidden to be accessed outside server
Number of Bytes per Second (by Sniffer)		❖	❖	❖	❖	❖	❖
Tune-in Delay (sec.)		❖	❖	❖	❖	❖	❖
Rebuffering Effect (sec.)		❖	❖	❖	❖	❖	❖
Remaining Observations & Corresponding Reasons		❖			❖		

CHAPTER TWELVE

RELATED WORKS

12.1 MBMS Streaming Performance Analysis

There are several standards specifications for MBMS streaming (3GPP, 2006) (3GPP, 2007). The work of Afzal (Afzal, Stockhammer, Gasiba, Xu, 2006) is the only main literature found on MBMS streaming to our knowledge. This paper evaluates and clarifies the impact of different parameters on the overall MBMS video streaming using H.264/AVC video coding and packetization performance over Enhanced General Packet Radio Service (EGPRS) and UMTS networks. In (Tian, Malakamal Vadakital, Hannuksela, Wenger, Gabbouj, 2005) the authors focus on the transmission of H.264/AVC video in the 3GPP MBMS. The proposed method which is simulated reduces tune-in delay and improves the quality of the video stream. This method requires scalable H.264/AVC coding and arranging of transmission order of the pictures according to their importance.

12.2 MPEG-7 based ESG Interface

The work of Hoyeon Jang (Jang, Moon, 2006) is about Integrated Electronic Service Guide (IESG) which is designed to compose ESG, interactive TV service information. The EPG concept is expanded to IESG. The channel program schedule is received over the air, the information of other channels and the genre are processed using a return channel. This work focuses on Advanced Common Application Platform (ACAP) software which is a Java based data broadcasting standard for interactive TV service through digital TV set-top boxes. This work covers unresolved issues for providing EPG service using Advanced Television System Committee (ATSC) Program and System Information Protocol (PSIP).

The work of Jihye Lyu et.al. (Lyu, Pyo, Lim, Kim, Lim, Kim, 2007) is about design of open APIs for Personalized IPTV that is a broadcasting service enabling support of interactive and personalized services using IP network. This study extends the Parlay X Web service structure for the personalized EPG, Network Data Recorder (NDR) and Target Advertisement services. The work covers personalized EPG and focuses on the personalized services.

In a recent study (Jin, Bae, Ro, Kim, Kim, 2006), Sung Ho Jin et.al. proposed an intelligent broadcasting system for enhanced personalized services, based on the semantics of the broadcasting content. In this study they used the MPEG-7 and TV-Anytime Forum (TVAF), as well as an agent technology. For content-level services real-time content filtering, personalized video skimming, and content-based retrieval using audio characteristic are implemented.

Andrea Kofler-Vogt et.al. in their study (Kofler-Vogt, Kosch, Heuer, Kaup, 2006) pointed out that multimedia delivery especially in mobile applications that deal with MPEG-7 suffer from limited bandwidth, low computational power, and limited battery life. So, they described an index system adopted from database systems that allow filter mechanisms and random access to encoded MPEG-7 streams and which overcome the limitation of the network and the consuming terminal. Encoding is applied in order to reduce the data rate of the XML documents to be transmitted. However, since MPEG-7 is a generic standard, not all components of MPEG-7 are necessary (Martinez, Koenen, Pereira, 2002) for mobile applications. Even, a small subset of MPEG-7 can meet basic requirements of mobile applications.

INSTINCT is a European project converging DVB-CBMS (DVB Convergence of Broadcast and Mobile Services) activities considering DVB-T (DVB Terrestrial), DVB-H, DVB-MHP (DVB Multimedia Home Platform). INSTINCT partners Brunel, FTR&D, Motorola and Netikos proposed an ESG data model which has been built over the skeleton of TV Anytime (TVA). That is, the ESG model is based on TVA and XML, uses MPEG-7 syntax to separate the content descriptions from the delivery instances. However, the proposed ESG introduces new features for IP

Datacasting over DVB-H. DVB-CBMS ESG model would be probably a compromise between TVA, Nokia and INSTINCT proposals while INSTINCT proposal is well perceived by DVB-CBMS group (Gross, Lauterjung, Levesque, Mazieres, 2005).

XML based ESG for mobile TV, EXPWAY's FastESG conforms mobile TV standards defined by DVB-CBMS, DVB-IPI (DVB Internet Protocol Infrastructure), TVA, W3C and MPEG groups, provides multimedia search engines, uses BiM (Cokus, 2005), an accepted binary XML transport format by MPEG-7, TVA, ARIB and DVB standards (Expway, 2007). EXPWAY uses the BiM compression method for audiovisual metadata. BiM has been adopted by the ARIB and TVA, and recently by DVB GBS (DVB Generic Data Broadcasting) and DVB CBMS for compression of EPGs and ESGs (Expway, 2007).

The IP Datacast over DVB-H integrates ESG which is based on MPEG-7 (ETSI, 2006) and TVA like existing standards. DVB Document A112 (DVB, 2007) discusses recommended restrictions on the use of imported MPEG-7 and TVA datatypes within the ESG data model.

CHAPTER THIRTEEN

EXPERIMENTS

13.1 Test Environment

13.1.1 The Usage Reason of NetEm Emulator

A network emulator is required to emulate wireless network conditions. An emulator includes both the simulation and the live testing of a real environment.

In this thesis study, NetEm (NetEm, 2006) Linux emulator has been used together with Linux bridging (Linux Bridging, 2007). NetEm can also be used with Linux acting as a router. All these tools are free. There are also other emulators, but the key factor determining why NetEm is used is that it can work with Ethernet frames. The experiments were upon multicast and broadcast networks. Any other emulators could not deal with the broadcast and multicast packets, since they only cared about IP packets. NetEm emulated network through Ethernet frames, so it did not make any difference between broadcast/multicast or a general IP packet. Linux Emulation host had two Network Interface Cards (NIC) attached. For working emulation on this Linux host, the host must copy packets from one network, apply emulation over them and then release to the other network. To enable this, Linux bridging was used, so bridge copied Ethernet frames from one network and released them to the other network. That directly allowed NetEm to copy the frames to its network queue. It is a protocol independent emulator which works over Ethernet frames. If frames are affected by the network delay and loss, the upper layer protocols such as RTP, RTSP (Real-Time Streaming Protocol), Transmission Control Protocol (TCP), UDP etc. are affected. NetEm was mainly used to test streaming quality under UMTS conditions.

13.1.2 Vidiator MBMS System Modules

Vidiator's MBMS Xenon Live Encoder, Xenon Service Manager and Xenon Streamer were used in this work to create MBMS streaming services (Vidiator, 2007) as shown in Figure 13.1.

Xenon Live Encoder, can encode both live and on demand video content in real-time and pass it to a streaming server for delivery. Xenon Live Encoder is being used by mobile operators and content providers for mobile multimedia services.

MBMS Xenon Service Manager, serves to create a new MBMS service. A service encapsulates delivery method parameters. Xenon Service Manager also shows existing MBMS services.

Xenon Streamer, is an MBMS streaming server. It is used to multicast or to broadcast the active streaming services. Xenon Streamer is connected to Xenon Live Encoder by Smart Live Session Protocol (SLSP) which is a Vidiator's protocol for communicating with Xenon Live Encoder.

Vidiator's MBMS File Delivery Over Unidirectional Transport (FLUTE) Download Server is used for MBMS download and announcement sessions that announce an occurence of a new user service by listing service details. The announcement service is a download service involving meta files sent to MBMS subscribers. Open Mobile Alliance (OMA) Electronic Service Guide (ESG) (OMA BCAST, 2006) is also provided for session announcement. A subscriber joins any one of the valid services through the Service Guide Receiver.

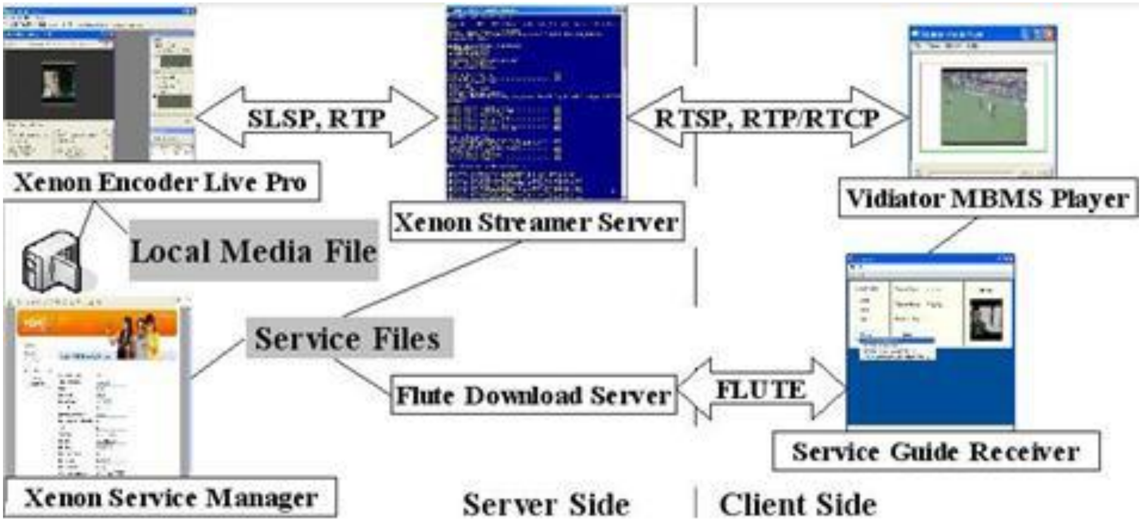


Figure 13.1 Vidiator MBMS system modules.

Figure 13.2 shows Vidiator's system model with OMA ESG. In this model, MBMS data is delivered through RTP (Real-time Transport Protocol) and FLUTE (File Delivery Over Unidirectional Transport) protocols, for streaming and downloading delivery modes, respectively.

As operators create a new service, OMA BCAST ESG files are created. Vidiator's Xenon Live Encoder encodes the media content. MBMS Broadcast Multicast Service Centre (BM-SC) uses MBMS delivery functions to transmit media and OMA BCAST ESG fragments to User Equipments (UE). The MBMS receivers download ESG meta files for accessing the relevant sessions. ESG and MPEG-7 meta files are transmitted over FLUTE protocol. Sessions are bundled groups of services in ESG. Several sessions are grouped according to the timing and genre criteria in the content server. Each criterion can be considered as a service. Service Guide Delivery Descriptor in administrative section is a root meta file that refers ESG fragment meta files.

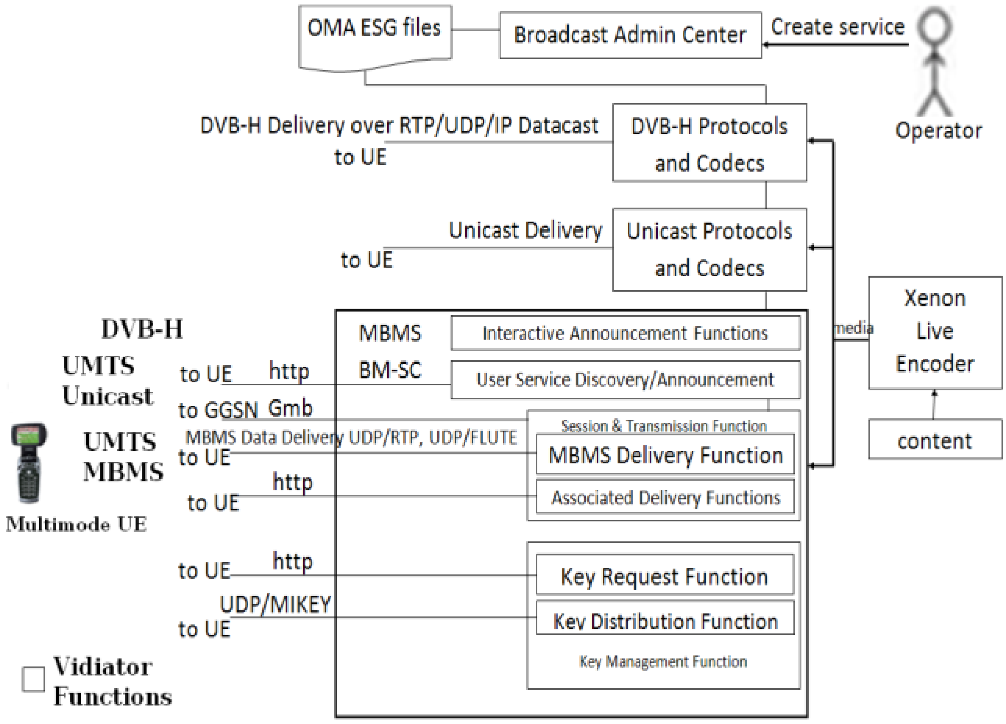


Figure 13.2 Vidiator system model with OMA ESG

13.2 Rebuffering Effect and Tune In Delay Results

3GPP TR 26.946 V7.0.0 Appendix A.2.1 and A.2.4 (3GPP, 2006) UTRAN streaming simulation conditions and results are used in this study, as in Table 13.2. Table 13.1 is about the characteristics of media that we used for all experiments as well as network conditions under evaluation. In our experiments, we focus on the observed and measured streaming quality improvement with and without FEC by emulating Local Area Network (LAN) as UMTS network on the following issues:

- Impact of packet loss (Service Data Unit, SDU)
- Bandwidth limitation impact (kbit/s)

We consider and analyze the following delays:

- Average tune-in-delay
- Rebuffering impact/delay

Tune-in-delay, which is the sum of protection period and decoding delay, experiments were repeated for every situation with respect to bearer rate (64, 128, 256 kbit/s) and Block Error Rate (BLER) (1, 5, 10%) using Reed Solomon FEC by given FEC overhead ratio values in Table 13.2 as well as using Null-FEC.

Rebuffering effect was measured as a sum of tune-in-delay, buffering time interval and short interrupt delays on video during the stream play-out. The short interrupt delays are considered as 500 milliseconds. The number of the interrupts are multiplicated by 500 milliseconds to get the delay summation of all short interrupts.

Average tune-in-delay (sec) values are given in Figure 13.3, Figure 13.4 and Figure 13.5 with the corresponding FEC overhead ratios. It is observed that with higher Reed Solomon FEC overhead ratio there is higher tune-in-delay for all the BLER 1%, 5%, 10% cases for 64 kbit/s bearer rate, as in Figure 13.3.

Table 13.1 Media characteristics

Video Codec	Video Bandwidth	Video Width, Height	Audio Codec (bit/s)	Audio Bandwidth	Audio Sampling Rate	Audio Channel Number
H.263	40446bit/s	176x144	AMR	12800 bit/s	8000 bit/s	Mono

Table 13.2 UTRAN network characteristics

Error Rates	Bearer Rate	MBMS FEC	FEC Overhead
Low 1% BLER	64 kbit/s	5 s: 55.8	12%
		20 s: 60.4	5%
	128 kbit/s	5 s: 115.5	9%
		20 s: 122.4	4%
	256 kbit/s	5 s: 236.2	7%
		20 s: 245.7	4%
Medium 5% BLER	64 kbit/s	5 s: 46.6	27%
		20 s: 53.6	16%
	128 kbit/s	5 s: 100.8	21%
		20 s: 111.8	12%
	256 kbit/s	5 s: 227.0	11%
		20 s: 224.0	12%
High 10% BLER	64 kbit/s	5 s: 38.5	39%
		20 s: 47.2	26%
	128 kbit/s	5 s: 87.5	31%
		20 s: 101.2	20%
	256 kbit/s	5 s: 179.5	29%
		20 s: 200.5	21%

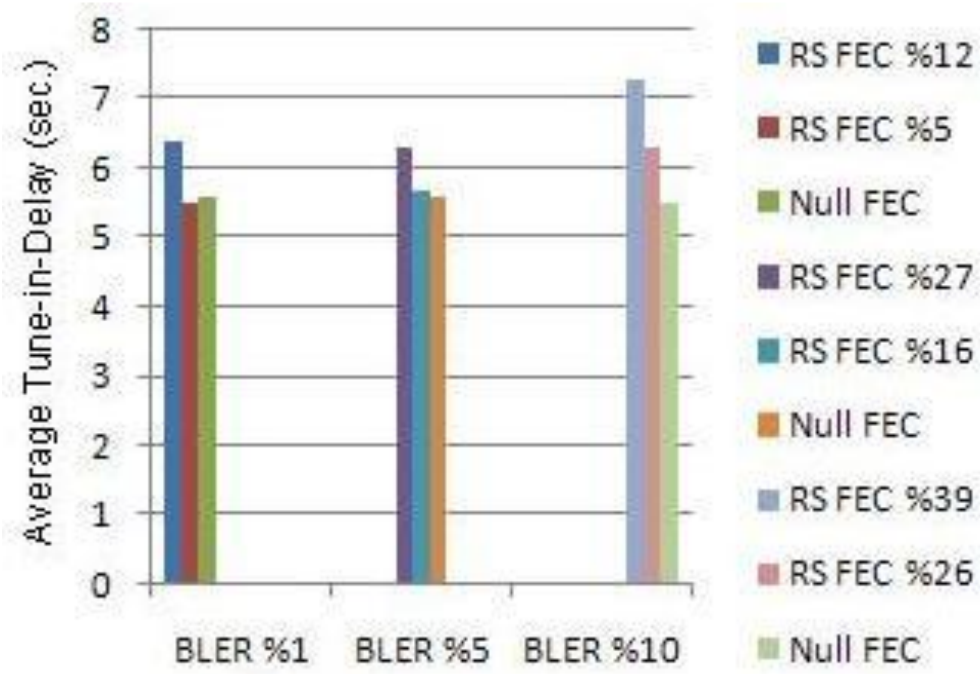


Figure 13.3 Average tune-in-delay results according to selected FEC overhead ratio with respect to bearer rate 64 kbit/s and BLER 1, 5, 10%.

Low tune in delay was observed with high FEC overhead ratio for 128 kbit/s bearer rate by 1%, 5% loss conditions. High tune in delay was observed by 10% loss condition with high FEC overhead, as in Figure 13.4.

High tune in delay was observed with high FEC overhead ratio for 256 kbit/s bearer rate by 1%, 5% loss conditions. Low tune in delay was observed by 10% loss condition with high FEC overhead, as in Figure 13.5.

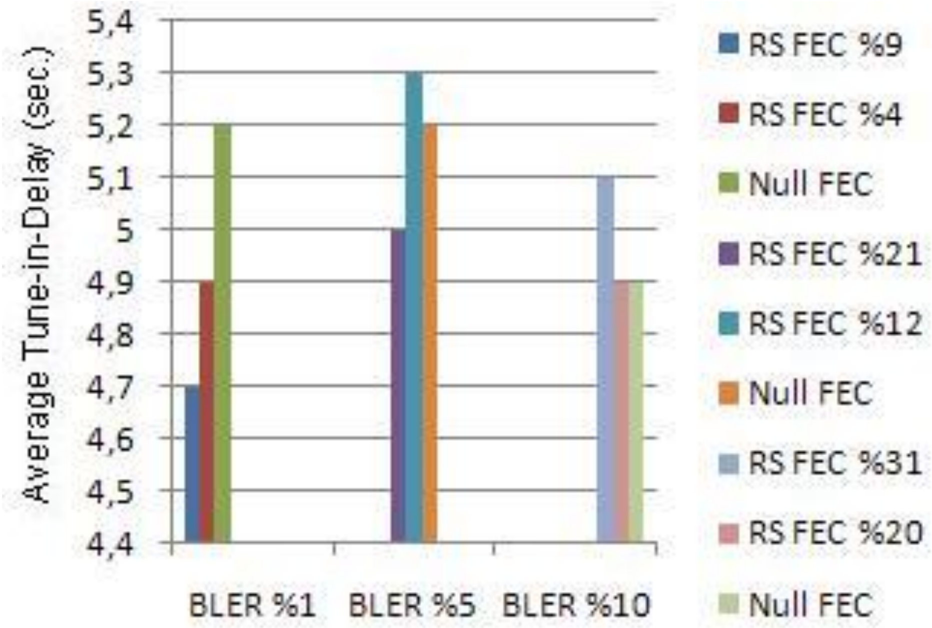


Figure 13.4 Average tune-in-delay results according to selected FEC overhead ratio with respect to bearer rate 128 kbit/s and BLER 1,5, 10%.

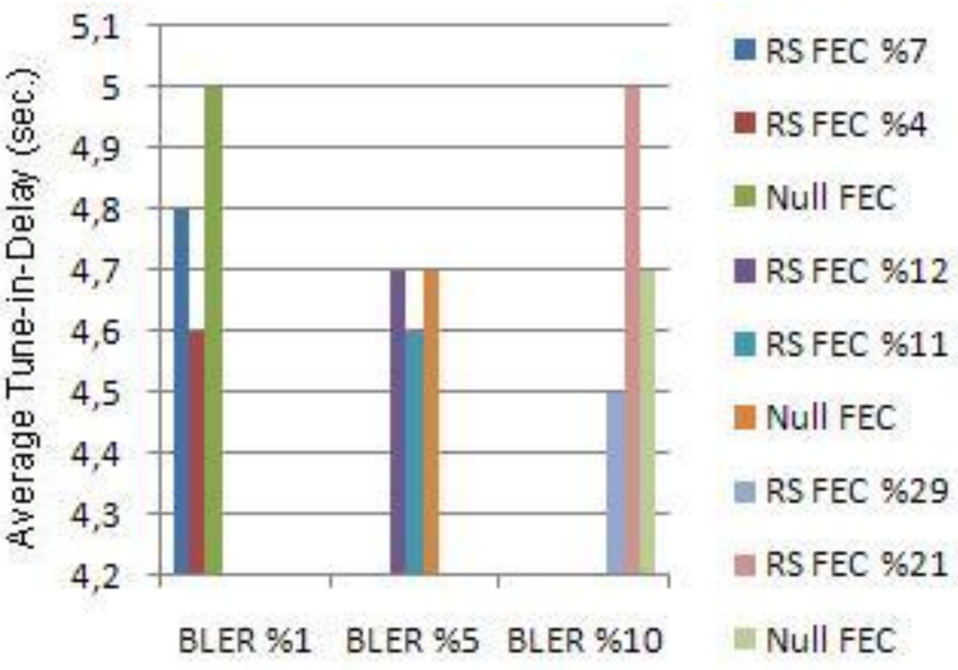


Figure 13.5 Average tune-in-delay results according to selected FEC overhead ratio with respect to bearer rate 256 kbit/s and BLER 1,5, 10%.

Total rebuffering effect results are given with Figure 13.6, Figure 13.7 and Figure 13.8. Low rebuffering effect was observed with high FEC overhead ratio for 64 kbit/s bearer rate by 1%, 5%, 10% loss conditions, as in Figure 13.6.

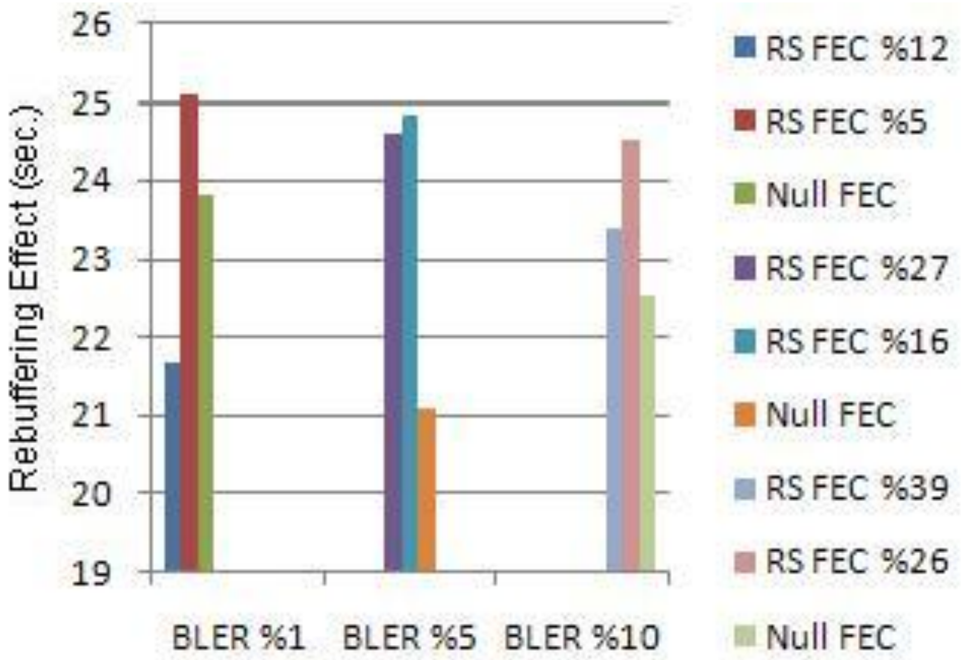


Figure 13.6 Total rebuffering effect results according to selected FEC overhead ratio with respect to bearer rate 64 kbit/s and BLER 1, 5, 10%.

Low rebuffering effect was observed with high FEC overhead ratio for 128 kbit/s bearer rate by 1%, 5% loss conditions. High rebuffering effect was observed by 10% loss condition with high FEC overhead, as in Figure 13.7.

Low rebuffering effect was observed with high FEC overhead ratio for 256 kbit/s bearer rate by 1%, 5% loss conditions. High rebuffering effect was observed by 10% loss condition with high FEC overhead, as in Figure 13.8.

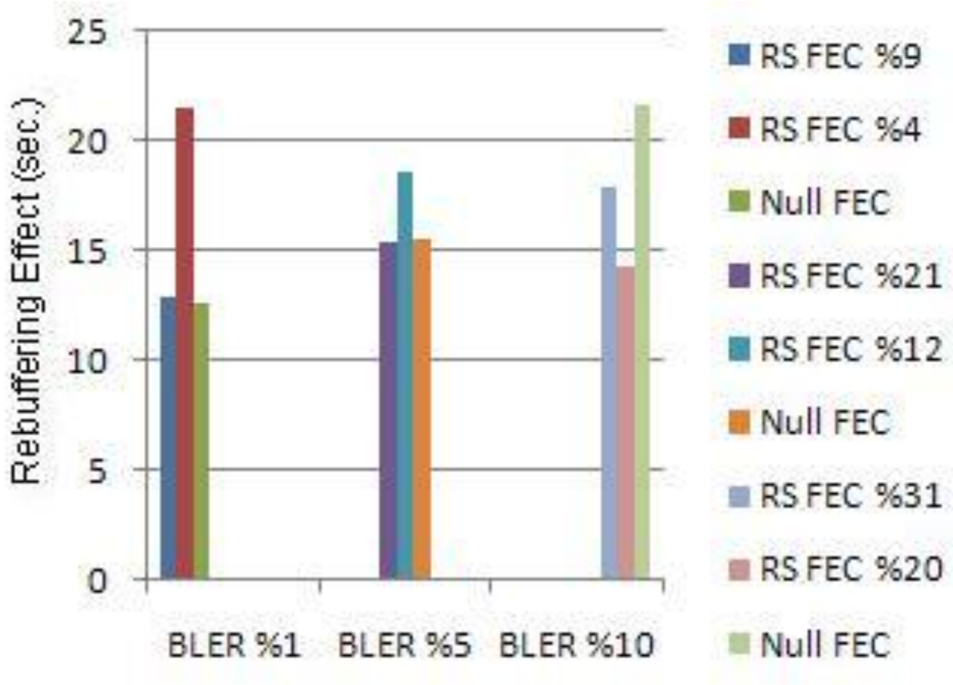


Figure 13.7 Total rebuffering effect results according to selected FEC overhead ratio with respect to bearer rate 128 kbit/s and BLER 1,5, 10%.

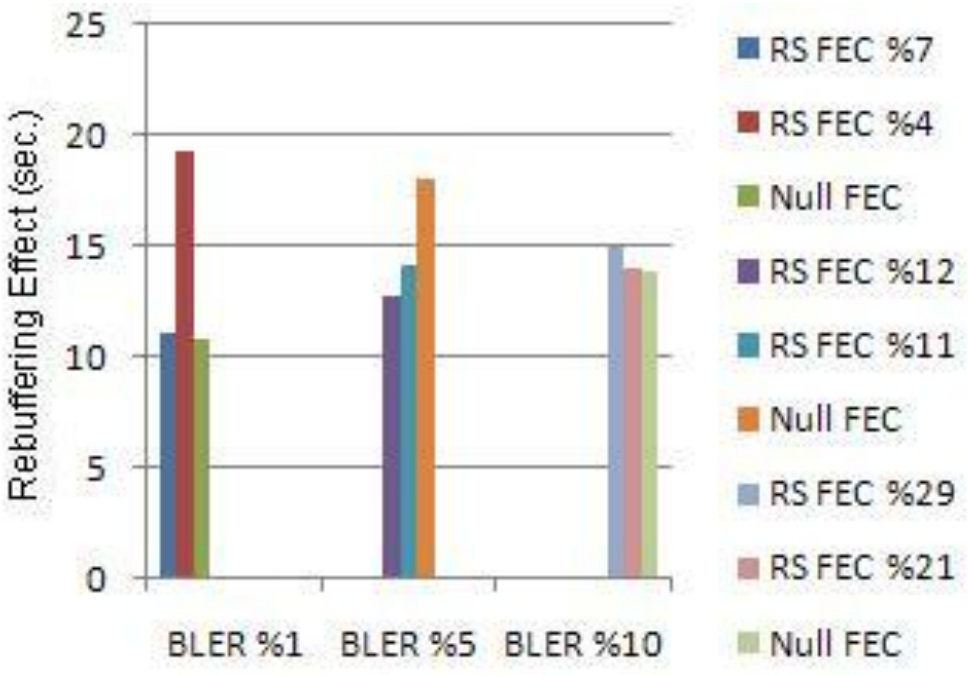


Figure 13.8 Total rebuffering effect results according to selected FEC overhead ratio with respect to bearer rate 256 kbit/s and BLER 1,5, 10%.

Table 13.3 summarizes the results of total rebuffering effect and tune-in delay experiments of which measurements are given by the figures above.

Table 13.3 Rebuffering effect and tune-in delay results

Bearer Rate	BLER	High FEC overhead ratio	
64 kbit/s	1%	Low Rebuf. Ef.	High Tune-in
	5%	Low Rebuf. Ef.	High Tune-in
	10%	Low Rebuf. Ef.	High Tune-in
128 kbit/s	1%	Low Rebuf. Ef.	Low Tune-in
	5%	Low Rebuf. Ef.	Low Tune-in
	10%	High Rebuf. Ef.	High Tune-in
256 kbit/s	1%	Low Rebuf. Ef.	High Tune-in
	5%	Low Rebuf. Ef.	High Tune-in
	10%	High Rebuf. Ef.	Low Tune-in

Video snapshots according to each used FEC overhead ratio values are given by Figure 13.9. It can be realized that FEC protects content due to network losses when compared to Null-FEC snapshots in Figure 13.10.

Figure 13.9 Reed Solomon video snapshots (Bearer rate kbit/s, BLER%, FEC overhead ratio%).

Figure 13.10 Null-FEC video snapshots (Bearer rate kbit/s, BLER%).

Raptor codes are more efficient for large “k”, source block lenghts. For large “k”, an additional cost of complexity occurs due to larger encoding and decoding matrixes. That means better code performance due to more loss recovery (Afzal, Stockhammer, Gasiba, Xu, 2006). Code rate “r” is equal to “k / n” ratio. Low code rate is equal to high FEC overhead ratio, as in Equation (1). At low code rates with low “k” or high “n” values, the number of repair symbols increases. That improves decoder performance, because the encoding and decoding matrixes are smaller. Reed Solomon algorithm complexity is given with $O(\log^2 k)$ globally. Reed Solomon totally is affected by a choice of “k”. If “k” is low, then the algorithm complexity decreases and decoder performance improves. According to our experimental results, rebuffering effect is lower when the FEC overhead ratio increases, because the decoder performance is better. All that means more inter symbol dependency.

In the case of high code rates, FEC protection is not enough to recover the channel losses. When the code rate decreases, the peak signal-to-noise ratio (PSNR), i.e. video quality, increases. The assumed and applied FEC overhead ratio values corresponding to the different bearer rate and BLER values show this quality improvement as shown in Figure 13.9. Even the network is lossy, the enough FEC overhead ratio, which means low code rate, is sufficient to receive error-free video which is a fact between Figure 13.9 and Figure 13.10. Raptor supports any desired tune-in-delay (3GPP TSG, 2004).

According to our results, tune-in-delay generally is higher with higher FEC overhead ratio. As given in (3GPP TSG, 2004), tune-in delay consists of the following parts:

1. Reception delay until the first packet of a FEC block increases when the number of FEC block symbols increases, since the total number of repair and source packets increases.
2. Reception duration of a complete FEC block depends on the size of source RTP packets and FEC block size according to the FEC scheme. At least one super block which is the minimal unit to random access to a stream must be ready before decoding.
3. Delay to compensate the size variation of FEC blocks.
4. Synchronization delay between the MBMS streams. Synchronization of video and audio streams is important for lip synchronization.
5. Delay until a media decoder is refreshed to produce correct output samples. The receiver should delay decoding of a FEC block until all its repair packets are received.

13.3 MPEG-7 based ESG Interface

In order to evaluate our approach we have developed a test application in Visual Studio .NET 2005 Smartphone Emulator, and a general view of application can be seen in Figure 13.11.

Vidiator's MBMS FLUTE Download Server is used for MBMS download and announcement sessions that announce an occurrence of a new user service by listing service details. The announcement service is a download service involving meta files sent to MBMS subscribers. OMA ESG (OMA BCAST, 2006) is also provided for session announcement. A subscriber joins any one of the valid services through the Service Guide Receiver.

Figure 13.11 Appearance of service guide test application.

The Service Guide Receiver (SGR) for mobile TV has been redesigned by extending MPEG-7. The SGR application has an iconized tree view of services, which are grouped according to genre under the content server source IP address. Iconized view also makes it easy to distinguish status of services as expired, streaming and scheduled services with different icons. Multimedia query interface, service details and its preview are shown in Figure 13.12-a, 13.12-b and 13.12-c, respectively.

The test application is also capable to perform simple multimedia query based on extended MPEG-7 that involves the content fragment in the core of ESG as in Figure 9.1. Multimedia query processor searches all the ESG files and MPEG-7 content descriptions. The query result grabs the service name, service genre by matching service to its location in a tree view structure and its status. If there is more than one possible result, these details make easier to assess the query result, as shown in

Figure 13.12-a. By selecting within the tree view service names or selecting one of the query results, a user can connect a streaming video service by connect service button.

Figure 13.12 Service guide receiver with multimedia query.

We have observed that with the integration of MPEG-7 the system performance and complexity were not impacted. The query response time has been very low. MPEG-7 has provided value-added features for a better user service guide usage.

CHAPTER FOURTEEN

CONCLUSION AND FUTURE WORK

14.1 MBMS Streaming Performance Analysis

MBMS streaming performance has been tested in this study. We considered packet loss and bandwidth limitation performance metrics for our experiments. As a general, we have observed that higher FEC overhead ratio increased tune-in delay and decreased rebuffering delay for the same bearer rates (64, 128, 256 kbit/s) and BLER (1, 5, 10%). Three exceptional results occured for the tune-in-delay with the bearer rate 128 kbit/s when BLERs were 1% and 5%, and with the bearer rate 256 kbit/s when BLER was 10% respectively. Besides two exceptional results occured for the rebuffering effect with the bearer rate 128 kbit/s when BLER was 10%, and with the bearer rate 256 kbit/s when BLER was 10%.

14.2 MPEG-7 based ESG interface

In this study we presented we propose MPEG-7 based Electronic Service Guide (ESG) based on Vidiator's MBMS architecture for Mobile TV applications. Our prototype covers OMA BCAST ESG fragments defined for MBMS and extends content fragment of ESG by MPEG-7.

We extended the existing OMA BCAST ESG with MPEG-7 meta data structure to provide a tree view and a genre based view of available services, and provided information on how service guide descriptions are addressed.

We have observed that the system performance and complexity were not impacted with the introduction of MPEG-7 meta data and the ESG experience has significantly improved with the addition of search feature which introduces negligible delays.

MPEG-7 contains a rich set of content description interfaces for multimedia data. However, limited resources of mobile environment force us to use a reduced version of MPEG-7. This is quite acceptable because MPEG-7 is a generic standard, not all components of MPEG-7 are necessary for mobile applications. In this study we have used only some of the manual annotations. However different description tools of MPEG-7, such as low level and some of the other semantic content descriptors can be used for mobile TV environment.

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