Multimedia Codec Evaluation and Overview

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ABSTRACT

Movies, songs, conferences, video streams and other multimedia services are spreading in everybody's life, needs and work. Different types of audio and video codecs of various qualities appear to be used for different purposes. However, each codec can be suitable for a special usage, as for example a codec that can be use for high quality film may not be suitable for streaming it into network. The main purpose of this thesis is to compare the performance of different codecs in one particular scenario and indicate which codec performance is the best.

The thesis work includes the investigation of several codec aspects (audio and video) and problems related to choosing a multimedia codec (coder/decoder) suitable for large-scale multimedia distribution over the Internet. More specifically, the work focuses on various features of modern media codecs, compares them and finds the best application for each of them. Furthermore, the thesis provides an overview of available research results related to this work and codec comparison tables for codecs and their features.

Additionally, as part of this work, a testbed was developed to measure the performance of codecs when used to stream media over a network. The testbed enabled the collection of various QoS parameters, such as bandwidth usage and burst, size for each codec considered in this thesis.

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CHAPTER 1
INTRODUCTION

Our modern life now is impossible to imagine without Internet. And with the passing of time the ubiquity of Internet is wide spreading everywhere, playing a greater and greater role in everyday life. As we see now streaming media and interactive events via Internet do not surprise anybody. Different types of media like speech, audio, video, text and images are integrated and used in a wide range of different subject areas. There are two certain key features of multimedia networks: they use high data transfer rates, and playback is performed in real time.

First of all, video and audio data are completely different in terms of their inner structure, owing to the fact they are created and played by different devices. In addition video media often includes subtitles in different languages. It is a common situation that a typical movie contains a large number of files: video files, several audio files, and several text files with subtitles in different languages. Moreover, if video and audio will be even slightly desynchronized it will immediately affect the quality of the movie. Also, there appear new problems that developers have to solve: what video codec is best suitable for particular type of video or audio, which multiplexer should be used when you combine video and audio, what container format to use etc. It becomes more difficult because a variety of codecs were developed for different purposes. A codec that one can use for high quality movie may not be suitable for streaming it into network. Since the network is characterized by limited bandwidth, one should definitely compress the file. For all compression systems two algorithms are necessary: one is the encoder at the source, and the other is the decoder at the destination. These two algorithms are usually asymmetrical. The film can be encoded just only one time before it will be placed on a server, while it can be decoded thousand times by users trying to see this film. Thus, the encoding algorithm could be complex and slow while decoding should be simple and quick. On the other hand a videoconference requires both algorithms to be quick and performed in real time. In addition, the encoding/decoding process can be irreversible, which means that you'll not necessarily get the exact copy of the original media file when decoding the encoded file. This allows achieving higher coefficient of compression.

This work is intended to investigate various multimedia video and audio codecs, using various encoding and streaming mechanisms, and evaluation of corresponding results.

1.1 OVERVIEW

Codec (short for coder/decoder) is the strategy of encoding /decoding (compression /decompression) of various types of data like music or video. It represents a compression algorithm to reduce the size of the stream. Codecs can be used for converting of analog video signals into compressed video files or analog sound signals into digitized form. Codecs are used either with streaming (live video or audio) or download (file-based) content.

There are different audio and video codecs. One kind of data forms elementary stream, which is encoded using a particular codec. Since any file in computer system is represented as a sequence of bits there has to be some way to retrieve the video or music in its natural origin. That's why the format is used to point out the system how the bit streams are organized, so that the operating system could correctly interpret the particular file. The container format may combine several elementary streams. Thus, media file format is one and it may encompass several codecs, for example audio codec and video codec in one multimedia file. One common method is to define file format by filename extension. Based on this operating system opens the file in appropriate container. Containers differ from each other on the base of file formats supported as well as support for additional capabilities.

Compression is the size reduction of the data file. The reduction of the data file is categorized into "lossless" and "lossy" compression. Both lossless and lossy compressions are used in data compression.

- The lossless concept is to encode the audio file with no discards of any of the information from the compressed data making the music or sound. The data produced are exactly the same original audio data. Lossless compression application is mainly in zip file format and may be used as a component within lossy data compression technology.

- Lossy compression is the contrast to the lossless compression where the reconstructed data is not exactly the same as the original source where the discarded information are the least important information.

1.2 THESIS OUTLINE

Our thesis report is built in the following way. It contains seven chapters. The current chapter describes the role of multimedia in our life and the way how it can be presented. Chapter two includes the overview and description of various audio codecs. Chapter three is mainly focused on the video codec overview. Chapter four is devoted to the experiment - tools, measurement infrastructure. Chapter five discusses the analysis of the measurements. Certain results are obtained for both bit rates 1024 and 3072 kbps. Chapter six gives a brief overview of the codec families based on the experiment and provides summary table for different codecs. Chapter seven outlines some key points for future work.

CHAPTER 2
AUDIO CODECS

Audio can be considered as a continuous series of air pressure waves. When audio is digitized, an analog recording is played back through the electronic device and variations of the electric current generated by the device are sampled at a very fast time intervals.
The quality and resolution of digitized audio is determined by two factors:

1. Sampling rate – the number of times per second the amplitude of the wave is measured.
2. Bit depth – range of numbers used to represent the measurements.

These two factors are considered with relation to the original audio source as well as with relation to the intended appliance.

Finally, the digital files can be recorded with different codecs format. Depending on its final designation the digital file can be encoded with different types of codecs. That can be uncompressed or lossless data format, which has high audio quality that is highly desirable for studio, backup or archive storage. The lossy data format is useful in case of some constraints like limited storage space or limited bandwidth.

The choice of the codecs is stipulated by its designation, popularity and widespread usage.
Thus, the following audio codecs are considered:

- Uncompressed format: WAV.
- Lossless data format: MPEG-4 ALS and WMA9.
- Lossy data format: AAC, MP3, WMA, Vorbis, and Real audio 10.

2.1 WAV

WAV is the abbreviation for Waveform Audio Format (sounds and music).

Its extension is ‘.wav’. It is a data format designed for computer use, and is compatible in different operating systems such as Microsoft Windows and Apple Macintosh (Mac OS) operating systems.

It contains mainly uncompressed audio in PCM (Pulse Code Modulation) format where the analog signals are represented in digital form, and the magnitude is sampled at uniform intervals. Then the obtained signals are represented in digital form. The WAV format gives maximum audio quality because PCM uses uncompressed lossless storage scheme and this allows maintaining all the samples of the audio signal. A drawback for the WAV audio codec is that the resulting files are large in size, which means large disk space consumption. However it can be unsuitable for transferring through the network due to limited bandwidth. There are other audio codecs resulting in smaller files size like MP3 and AAC.

The wav format can handle up to a 4GB file size equivalent to 6.6 hours of CD audio (44.1 kHz, 16-bit stereo). WAV is outlined by its maximum audio quality in addition to its compatibility in different operating systems. One main drawback for WAV is that WAV files are big in size.

2.2 MPEG-4 ALS

MPEG-4 ALS is the abbreviation for Moving Pictures Enhanced Group 4 Audio Lossless.

MPEG-4 ALS is an extension of the MPEG 4 audio coding family. ALS provides methods for lossless compression of audio signals with any sampling rates, up to 2^{16} channels and its resolution is up to 32 bit. Its operation is similar to FLAC (Free Lossless Audio Codec).

MPEG-4 ALS data packing is efficient for audio data. It allows getting high quality records at significantly reduced data rates. MPEG-4 ALS mainly uses linear prediction coding (LPC), Golomb Rice Coding and run length encoding (RLE) for its operation.

LPC can be considered as subset of filter theory to convert the audio samples to a sequence of small samples. It is used to represent the spectral envelope of a digitized audio signal in compressed form. Golomb Rice Coding is a scheme to convert codes to symbol where the small audio samples converted by LPC are stored using Golomb Rice coding. RLE is a very simple form of data compression in which sequences of repeated data are stored in a single data value and a count, rather than in the original form.

MPEG-4 ALS ensures that an exact duplicate of the original data is created which is a good benefit for using it. It is efficient and provides fast lossless audio compression techniques for consumer and professional usage.

The important benefit of MPEG 4-ALS coding is that it provides some features not available to other lossless compression formats:

- Support for any uncompressed digital audio format.
- Support for PCM resolutions of up to 32-bit at any sampling rates.
- Support for up to 2^{16} channels.
- Optional storage in MPEG-4 file format.

2.3 WMA 9 Lossless

WMA 9 Lossless is the codec abbreviation for Window Media Audio 9.

Its extension is '.wma' and it is considered as audio file format. It is a lossless audio codec with samples audio of 96 kilohertz (kHz) using 24bits that makes it ideal for archive or backup storage. The data compression ratio is 2:1 or 3:1 depending on the complexity of the source.

It is compatible with different OS and backward compatible with previous WMA standards. In addition that it has Digital Rights Management platform included (copy-protected digital media).

2.4 MPEG

MPEG stands for Movie Picture Expert Group (RFC3003).

MPEG Audio it is a family of three audio codecs and compression schemes:
Layer 1, Layer 2, Layer 3. The complexity of the algorithm increases with the number of layer, but that doesn't guarantee the highest quality in all the cases. The compression technique used by MPEG is a combination of transform-encoding and sub-band division. MPEG-Audio operates at a low bit rate (from 32 to 448Kbps per monophonic channel) encoding with sampling rates 32, 44.1 or 48 kHz.

2.4.1 MPEG-1

MPEG-1 is a simplified version of MUSICAM. (Masking-pattern Adapted Universal Sub band Integrated Coding and Multiplexing). MUSICAM is a primary audio format used for digital video broadcast and direct satellite system. MPEG-1 operates pretty well at 192 or 256Kbps per channel.

2.4.2 MPEG-2

MPEG-2 is identical to the MUSICAM standard. It is optimized for a bit rate of 96 or 128 kbs per monophonic channel. In stereophonic mode MPEG2 is nearly equivalent to CD quality [12]. MPEG-2 AAC is a high quality multichannel audio coding system, which is used for HDTV, DVD, and cable and satellite television. It encodes multiple channels of audio into a low bit-rate format. High compression rate is achieved due to encoding of multiplicity of channels as a single entity.

2.4.3 MP3

MP3 is MPEG-1 Audio Layer 3. It is the highest performing scheme from standard MPEG series. More complex encoding scheme allows achieving nearly CD quality at 64Kbps per channel. It supports a great variety of bit rates in addition to multiple compression levels.
The quality of MP3 files depends on the quality of the encoder and the complexity of the encoded signal.

In addition to the availability of wide range of bit rate and frequencies, Variable and non-standard bit rates are also possible, however, no DRM protection technology, but patented.

The MP3 limitations are the following: bit rate is limited to a maximum of 320 kbps, time resolution can be too low for highly transient signals, joint stereo is on a frame-to-frame basis, encoder/decoder overall delay is not defined.

2.5 AAC

The AAC abbreviation means Advanced Audio Coding in MPEG-2 Part 7 and MPEG Part 3.

This codec provides better sound quality relative to the bit rate because the irrelevant signal components are discarded and redundancies in the coded signal are eliminated. The signal is processed by a modified discrete cosine transform (MDCT) algorithm according to its complexity. Though the standard is rather new, no incompatibilities are discovered.

AAC has the following multiple codecs:
- Low Complexity Advanced Audio Coding (LC-AAC).
- High-Efficiency Advanced Audio Coding (HE-AAC).
- Scalable Sample Rate Advanced Audio Coding (AAC-SSR).
- Bit Sliced Arithmetic Coding (BSAC).

AAC is characterized by its low delay in network latency: 20 ms and a good audio quality for all kind of audio signals. It encompasses error protection (EP) tool and error resilience techniques.

Although AAC requires patent license for manufacturing or developing of AAC codec, it is not a proprietary format and no payments are required to stream or distribute files in AAC format.

2.6 WMA

Abbreviation stands for Windows Media Audio. It is a proprietary codec of Microsoft.

WMA9 codec samples audio at 44.1 or 48 kHz using 16 bits with the data rates from 64 to 192 kbps. It supports CBR and VBR. With the development of WMA10 Pro a lot of features were added to enhance the functionality of this codec. It can support 24- bit/96kHz stereo, 5.1 channels or even 7.1 channels surround sound. WMA10 Pro offers streaming, progressive download, download-and play delivery at 128 to 768Kbps. This makes it applicable for wide range of playback devices and methods. WMA10 Pro is patented and supports DRM. Verification of performance results was conducted by National Software Testing Labs.

2.7 VORBIS

Vorbis is a free and open source lossy audio compression codec developed by Xiph Foundation. The development of Vorbis was stimulated by announcement of MP3 licensing. That's why Vorbis was quickly adopted by the market. Many video games and consumer electronics audio are stored in Vorbis format.

At standard input at 44.1 kHz encoder produces digitized output sequence from 32kbit to 500kbit, with VBR and different quality settings. It uses a Modified Discrete Cosine Transform for converting data from time domain into frequency domain. After that data are broken into noise floor and residue components then quantized and entropy coded using a codebook based vector quantization algorithm [6].

Vorbis supports bit rate peeling and support for metadata (here they are called comments).

2.8 AC3

AC3 – Acoustic Coder 3 is a high-quality audio codec (audio coding format) elaborated by Dolby Laboratories.

AC3 achieves large compression ratios by encoding multiple channels of audio into a low bit-rate, and encoding a multiplicity of channels as a single entity format. It uses a hybrid backward/forward adaptive bit allocation approach, which is necessary for advanced television [11].

Dolby Digital – one of the versions of AC-3 encodes up to 5.1 channels of audio.AC-3 has been adopted as an audio compression scheme for many consumer and professional applications. It is a mandatory audio codec for DVD-video, Advanced Television Standards Committee (ATSC), digital terrestrial television and Digital Living Network Alliance (DLNA), home networking, as well as an optional multichannel audio format for DVD-audio.

2.9 RealAudio 10

RealAudio10 is proprietary codec of Real Networks. Its range of scaling is from 12 to 800kbps. It provides extremely high audio quality of sound at the widest possible bandwidth range.

To achieve such a superior quality the Real Audio codec is used for bit rates that are less than 128kbps, but at bit rates higher than 128kbps it incorporates AAC codec.
The original data from audio spectrum are divided into distinct frequency bands and that ones, which are imperceptible by human ear, are discarded. This allows achieving small file size with no noticeable degradation in sound.

Real Audio is highly widespread in portable and mobile devices market as well as in streaming, on-demand and download solutions.

CHAPTER 3
VIDEO CODECS

Video refers to representing of sequences of images to create motion scenes. Video nowadays features entertainment, interpersonal and interactive applications. There are several basic characteristics that define the choice of codec, format and container.

- Frame rate defines the number of frames per second. There are different systems like PAL, NTSC, SECAM, depending on the country.
- Scanning sequence refers to the process of refreshing the screen. It can be progressive or interlaced.
- Color signals include luminance and chrominance characteristics. Luminance refers to brightness of the source, while chrominance corresponds to hue and saturation.
- Digitization format depends on the application purpose. There are different formats like 4:2:0, 4:2:0, SIF, CIF, and QCIF.
- Resolution defines the size of the video screen and is measured in pixels for digital video or in horizontal scan lines for analog video. Aspect ration corresponds to the dimension of video screen and video elements. It can be 1:1, 3:4, 16:9 and so on.
- Reduction of the size of video can be done with interframe and intraframe compression.
- Video streams can be encoded with a particular rate, which refers to a bit rate. It can be constant or variable.
- Video quality depends on capturing method and used storage, and can be defined by subjective video quality or PSNR, where PSNR refers to peak signal-to-noise ratio, which represents the ratio of maximum possible power of a signal to power of corrupting noise.

The following video codecs are considered in this chapter:
H263v2, H264, MPEG-1, MPEG-2, MPEG-4, DIVX, XVID, Theora, Real video, and WMV.

3.1 H.263.v2

This is a low bit rate encoding solution for videoconferencing. It was developed by ITU-T as an evolutionary improvement over H.261 standard. It is superior to H261 at all bit rates. H263 is based on CIF, but allows using a wide range of custom source formats. It uses motion compensation capability as well as inter-picture prediction to utilize temporal redundancy and remove spatial redundancy. It is capable of CBR and VBR. H263 has improved slice structure and PB-frames mode.

It provides symmetry of transmission, which means that codec can be used for bidirectional or unidirectional communication. This makes this codec very suitable for video telephony. It also provides up to eighteen negotiable coding options for improved performance and increased functionality.

H263 is patented. With the development of more enhanced standard H.264 the usage of H.263 significantly decreased [19].

3.2 H.264/MPEG – 4 AVC

Advanced Video Coding – the product of a collective partnership between VCEG (Video Coding Expert Group) and MPEG (Joint Video Team) – is a digital video codec standard that aims to achieve very high data compression. The standard can be applied to a wide variety of applications: low and high bit rate, low and high resolution video thus different types of networks and systems: broadcast, DVD, RTP/IP etc.

AVC includes 7 profiles:

- Baseline Profile (BP) - videoconference and mobile applications - lower cost applications demanding less computing resources.
- Main Profile (MP) – broadcast and storage application.
- Extended Profile (XP) – streaming – high compression capability.
- High Profile (HiP) – disc and storage application primarily High-definition television applications like HD DVD, Blu-ray Disc.
- High 10 Profile (Hi10P) -top of High Profile – adds up to 10bps of decoded picture precision, which implies higher sharpness.
- High 4:2:2 Profile (Hi422P) up on Hi10P -professional applications including interlaced video add support for 4:2:2 chroma sampling while using up to 10bps of decoded picture precision. Chroma Sampling is used in many video encoding schemes to allocate more bits for luminance than for color information.
- High 4:4:4 Profiles (Hi444P) -deprecated.

Some subsets to the original standard have been added known as Fidelity Range Extensions:

- Higher fidelity video coding by increased sample accuracy (10-bit and 12-bit coding).
- Higher resolution color information (YUV 4:2:2 and YUV 4:4:4).
- 4x4 and 8x8 integers transform.
- Efficient inter picture lossless coding.
- Support of additional color spaces.
- Residual color transforms.

New features for more effective compression include [5]:

- Multi-picture motion compensation allows up to 32 reference pictures (instead of 1 or 2) that improves bit rate and quality in most scenes.
- Variable block size motion compensation (16x16 or 4x4) leads to more precise segmentation.
- Macroblock pair structure (not supported in all profiles), allowing 16x16 macroblocks in field mode (vs. 16x8 half-macroblocks in MPEG-2).
- Data partitioning (DP), a feature providing the ability to separate more important and less important syntax elements into different packets of data, enabling the application of unequal error protection (UEP) and other types of improvement of error/loss robustness (not supported in all profiles).
- Supplemental enhancement information (SEI) and video usability information (VUI), which are extra information that can be inserted into the bit stream to enhance the use of the video for a wide variety of purposes.
- Frame numbering, a feature that allows the creation of "sub-sequences" enabling temporal scalability by optional inclusion of extra pictures between other pictures, and the detection and concealment of losses of entire pictures (which can occur due to network packet losses or channel errors).

H264 is patented, though there is X264 codec, which represents the GPL version of H264.

3.3 MPEG

MPEG is the abbreviation for Movie Picture Expert Group.

MPEG is created by ISO, to standardize set of applications, which involves video with sound. Standards are classified by three parts: video, audio and system. Video and audio are concerned with the way how each bit stream is compressed and formatted, while the system part is responsible for integrating of two streams together for production of synchronized output stream [9].

We considered the following MPEG standards:

- MPEG-1, a standard for storage and retrieval of moving pictures and audio on storage media.
- MPEG-2, a standard for digital television.
- MPEG-4, a standard for multimedia applications.

3.3.1 MPEG-1

This video standard uses similar video compression as H.261. It uses Source Intermediate format (SIF) that means obtaining a picture quality compared to a video cassette recorder (VCR). It uses sub sampling technique with temporal resolution that gives frame refresh rate 30Hz for 525-line system and 25Hz for 625-line system.

MPEG-1 uses non interlaced scanning with digitizing format 4:1:1. The standard allows using I-frames only, I and P frames only, or I, P, B frames. D-frames are not supported, so therefore for usage of control functions with such video I frames should be used. If we assume the maximum allowed delay time 0.5 sec that factor will define the maximum separation of I frames in our frame sequence.

As to the compression algorithm used, it is similar to H.261 with some differences. Time stamps are can be inserted within a frame that so in case of damaged or missing macroblocks the decoder can quickly resynchronize. Appearance of B-frames increases the time intervals between I and P frames. Finer resolution is used to improve the accuracy of moving vectors.

MPEG-1 is the most compatible format in MPEG family, suitable almost for all computers and VCD/DVD players with acceptable quality at 1.5Mbs and resolution 352x240. However MPEG-1 supports only progressive video, which reduces motion smoothness, and it has half frame rate of interlaced scanning.

3.3.2 MPEG-2

The solution to the MPEG-1 disadvantage was found in MPEG-2 that supports both interlaced and progressive scan video streams. In progressive scan streams, the basic unit of encoding is a frame, while in interlaced streams the basic unit may be either a field or a frame.

MPEG 2 specifies a video stream format which may be constructed of three types of frame data (intra frames, forward predicted frames and bidirectionally predicted frames) that can be arranged in a specified order called the GOP structure (Group Of Pictures) [9].

A number of levels and profiles have been defined for MPEG-2 video compression. Each of these describes a useful subset of the total functionality offered by the MPEG-2 standards. An MPEG-2 system is usually developed for a certain set of profiles at a certain level. Basically:

- Profile = quality of the video.
- Level = resolution of the video.

MPEG 2 supports 4 levels (low, main, high 1140 and high) and 5 profiles associated with each level (simple, main, spatial resolution, quantization accuracy and high).
Since the low level of MPEG-2 is compatible with MPEG-1 the most attention deserves main profile at the main level (MP@ML) and two high levels related to HDTV.
Most decoders also support MPEG-1.
Because the MPEG-2 standard provides good compression using standard algorithms, it has become the standard for digital TV. Its features are:

- Video compression which is backwards compatible with MPEG-1.

- Full-screen interlaced and/or progressive video (for TV and Computer).
- Enhanced audio coding (high quality, mono, stereo, and other audio features).

The list of systems which use MPEG-2 is quite extensive and continuously growing: digital TV (cable, satellite, terrestrial broadcast), Video on Demand, Digital Versatile Disc (DVD).

3.3.3 MPEG-4

Main application of MPEG-4 standard is interactive multimedia application over the Internet. Standard expands accessibility options allowing not only passively access the video sequence like stop/play/pause but also to manipulate individual elements within the scene. High coding efficiency allows it to run over a low bit rate networks.

First of all MPEG-4 has a number of content-based functionalities, that means that before compression each scene is defined as a background with several audio-video objects (AVOs). Each AVO in its turn is defined by one or more video and/or audio objects. Each video or audio object has an object descriptor and can be manipulated through BIFS language, which is a language used to modify and describe the objects. Binary Format for Scenes is a format for audiovisual content. It is based on VRLM and is used to modify and describe the objects. With its help various actions with an object can be done, like delete, change the shape or color of an object, change its appearance or animate it. In addition, at a higher level the composition of the scene in terms of AVO is defined in separate scene descriptor.

Thus the compressed audio and video information related to each AVO is elementary stream which is transmitted over the network in form of transport stream. A number of techniques are included into MPEG-4 to make it more resilient to transmission errors. That is:

- Use of fixed length video packets as the base level data structure instead of GOBs.
- New variable length coding (VLC) scheme based on reversible VLC.

Visual features of MPEG-4 can be summarized as follows:

3.3.4.1 Formats Supported

The following formats and bit rates are be supported by MPEG-4 Visual:

- Bit rates: typically between 5 kbps and more than 1 Gbps.
- Formats: progressive as well as interlaced video.
- Resolutions: typically from sub-QCIF to 'Studio' resolutions (4kx4k pixels).

3.3.4.2 Compression Efficiency

- For all bit rates addressed, the algorithms are very efficient. This includes the compact coding of textures with a quality adjustable between acceptable for very high compression ratios up to near lossless.

- Efficient compression of textures for texture mapping on 2-D and 3-D meshes.

3.3.4.3 Content-Based Functionalities

- Content-based coding of images and video allows separate decoding and reconstruction of arbitrarily shaped video objects.
- Random access of content in video sequences allows functionalities such as pause, fast forward and fast reverse of stored video objects.
- Extended manipulation of content in video sequences allows functionalities such as warping of synthetic or natural text, textures, and image and video overlays on reconstructed video content. An example is the mapping of text in front of a moving video object where the text moves coherently with the object [9].

3.3.4.4 Scalability of Textures, Images and Video

- Complexity scalability in the encoder allows encoders of different complexity to generate valid and meaningful bit streams for a given texture, image or video.
- Complexity scalability in the decoder allows a given texture, image or video bit stream to be decoded by decoders of different levels of complexity.
- Temporal scalability allows decoders to decode a subset of the total bit stream generated by the encoder to reconstruct and display video at reduced temporal resolution. A maximum of three levels are supported.
- Quality scalability allows a bit stream to be parsed into a number of bit stream layers of different bit rate such that the combination of a subset of the layers can still be decoded into a meaningful signal. The bit stream parsing can occur either during transmission or in the decoder. The reconstructed quality, in general, is related to the number of layers used for decoding and reconstruction.
- Fine Grain Scalability is a combination of the above in fine grain steps, up to 11 steps

3.3.4.5 Other issues addressed by MPEG-4 Systems

- A standard file format that supports the exchange and authoring of MPEG-4 content.
- Transport layer independence. Mappings to relevant transport protocol stacks, like (RTP)/UDP/IP or MPEG-2 transport streams can be or are being defined jointly with the responsible standardization bodies.
- Text representation with international language support, font and font style selection, timing and synchronization.
- Datasets covering identification of intellectual property rights relating to media objects.

As we see, MPEG-4 absorbs many of the features of MPEG-1 and MPEG-2, also new ones like extended VRML support for 3D rendering, object oriented composite files (including audio video and VRML object), support for Digital Rights Management.
It is a flexible standard because of its multiple levels and profiles, so the decision about how to implement the features is left to the developer.

MPEG-4 can be used for any field, which benefits from compressing of audio/video stream like web (streaming) or CD distribution, videophone or broadcast TV.

MPEG-4 is patented. So even having readily available MPEG-4 video the license is still needed to use it legally.

3.4 DIVX

DIVX is a video codec created by DIVX.Inc which is popular due to its ability of compressing lengthy video segments into small sizes while still maintaining a high visual quality. It is based on lossy MPEG-4 part 2 which means:

- Support for MPEG style quantization
- Support for interlaced video
- Support for B-frame
- Quarter-sample and global motion compensation

There are two basic types of compressed frames: intracoded frames called I-frames and predictive frames which can be two types: predictive or P-frames and bidirectional or B-frames. B-frames provide high level of compression and they don't propagate errors because they are not involved in the coding of other frames. Provided with its own container, DIVX allows to record DVD movies. This causes a lot of controversy due to its usage in replication and distribution of copyrighted DVDs.

Supports the following DVD like features

- Interactive video menus
- Multiple subtitles
- Multiple audio tracks
- Chapter points
- Multiple formats
- Other metadata

From July 2004 no further DIVX software includes adware which was accompanying this codec software before (spyware). DIVX codec and DIVX are available for free but additional features require payment. It is closed source, but the open source version
DIVX is licensed, though open source version was released in 2001 which served as a base for Xvid [16].

3.5 XVID

Xvid is Free Software under the GNU Public license open source version of MPEG-4 codec [7]. It uses Advanced Simple Profile that can be characterized as:

- Support for MPEG-style quantization
- Support for interlaced video

- Support for B-frames
- Quarter-sample motion compensation (QPel)

The global motion compensation feature is not actually supported in most implementations although the standard officially requires decoders to support it.
Xvid is considerably fast in de- and encoding. It provides extremely high compression ratios and relatively good picture quality. There are a lot of applications supporting Xvid playback and no spyware added.

Compared with DIVX it should be noted that while DIVX is closed source and runs only on Windows, MacOS and Linux, Xvid is open source and can be potentially applicable for any platform.

3.6 Real Video

Real Video (RV) is a proprietary video codec created by Real Networks. It is supported by many platforms like Windows, Mac, Linux, Solaris and some mobile phones.
In pair with Real Audio which is supplied in Real Media container, it allows streaming media since it doesn't require downloading the video in advance.

The first versions of Real Video were using the H.263 codec but with release Real Video 8 company switched to their own proprietary codec. This codec is defined by the four character codes RV10 and RV20 of H.263 and RV30 and RV40 are Real Networks proprietary formats.

RV uses RTCP for setting up and managing the connection. But actual data transmission is done through proprietary RDT protocol.
RV is suitable for both constant bit rate and variable bit rate encoding which allows to achieve better video quality, however variable bit rate is less suitable for streaming as it is difficult to predict the capacity of a certain video stream.
RV doesn't record Real Video streams. This feature is quite useful for film studios, music labels and broadcasters because it is exclude the opportunity of dumping the video onto user's machine [15].

3.7 WINDOWS MEDIA VIDEO

WMV is a generic name for a family of video codecs created by Microsoft [8]. Originally it was a proprietary codec for low bit rate streaming, but with Windows Media Video 9 it became a standard. WMV used MPEG-4 Part 2 but with standardization of WMV 9 it became an independent unique codec.
WMV is usually packed into Advanced System Format container (ASF). This format doesn't

specify how the video should be encoded, but instead specifies the structure of video/audio stream. The ASF container structure is patented in the USA though the format is published; the license limits it to closed-source development projects only. Encapsulated in ASF format, WMV can support DRM protected content.

WMV is usually streamed on the net using Microsoft's proprietary MMS protocol. WMV can be used not only for distribution of video over the Internet but also for HD DVD.

3.8 THEORA

This is a lossy video compression codec developed by Xiph.org Foundation. It is based on VP3 codec whose quality and bit rate can be comparable to MPEG-4 Part 2 video. Unlike VP3, which is patented Theora is royalty-free and can be used for any purpose [6].
Though this codec is still under the development (there are just Alpha for libtheora: releases Alpha 1 – Alpha 7).
Actually, it is used with Vorbis audio codec in Ogg container format but compressed video can be stored in any suitable container format.

CHAPTER 4
EXPERIMENTS

4.1 DESCRIPTION

To meet our goals and construct the table of codecs, we had to prepare necessary test beds and run an experiment. We constructed a small network consisting from two Windows computers and one Linux computer. The Windows computers were functioning as a client server model while the Linux computer was used to capture the packets. For streaming purposes, on both Windows machines, we used VLC. The choice of VLC was stipulated by the following reasons:

- VLC is compatible on multiple platforms.
- It doesn't require to be configured on servers or clients side.
- It has built-in encoding and decoding facilities.
- A large variety of video and audio codecs, bit rates and formats are supported.
- Streaming can be done using a graphical interface as well as the command line.

The movie (The 5th Element) that we used for encoding was two hours two minutes long:

- DVD format,
- 850 seconds long,
- Multimedia file with MPEG2 video,
- The resolution was 720x576 (4:3),
- The bit rate is 8400 kbps,
- Audio Dolby AC3 with 448 kbps.

Using VLC, we extracted 14 minutes 10 seconds interval, which we considered most suitable for streaming. The first six minutes contained quiet scenes while last eight minutes contained active scenes. Since encoding can be done with variable frame rate, which means less bits are allocated for encoding low motion scenes (each subsequent frame doesn't differ too much from the previous one), than for encoding fast motion scenes (subsequent frames are much different). This will allow investigating the codec capabilities closer and detaily.
The next step was to encode this extracted part using different available video and audio codec combinations. For the encoded combination, the appropriate format is MPEG transport

stream (MPEG TS) because it is the most suitable format for transferring through the network. For transcoding in VLC for a file called test.wmv for example; the following command was used:

vlc –vvv test.wmv --sout '#transcode {vcodec=(videocodec) acodec=(audiocodec), vb=(videobitrate) ab=(audiobitrate): std{access=file, dst=(path to save the file)/test.ts}}' vlc – quit

This command takes the initial file *test.wmv*, encodes it using the chosen video and audio codec, at the chosen video and audio bit rate, and saves it as *test.ts*. In our experiment, we choose two video bit rates 1024 kbps and 3072 kbps, and audio bit rate was 192 kbps.
After having transcoded the movie in all the available combinations, we have to stream from server to client using the following VLC command:

- For server:
  vlc (filename) - - sout udp: (ip address of the client)

- For client:
  vlc –vvv udp:

Using these two commands, we were able to stream the prepared transcoded samples from server to client.

Meanwhile, the third computer was using tcpdump program to capture all the packets that were directed from server to client, using the following commands:

sudo tcpdump
tcpdump dst (clients ip address) –w (filename.pcap)

First of all, all our captured packets were saved as pcap files to allow the replay of the experiments values.

After that, we are interested in dumping our files into some text format without any unnecessary information. Tcpdump procedure gives us timestamp which is the time when the packet was captured, length of the packet and the protocol used. That is the exactly info that we need for other future calculation.

sudo tcpdump
tcpdump –r (filename).pcap –tt –l –n –q >(filename).txt

Now we got the testbed for our jitter, burst and bandwidth analysis. But before, we should have sorted our output in accordance to the corresponding goals. For that purpose, we used small Perl scripts. The results were saved as an .asc file in order to run the output in matlab and get the plots and histograms for our analysis.

4.2 MEASUREMENTS

For conducting of our measurements and analysis we need software, which could capture the packets and provide us with the necessary data. Tcpdump provides us such capability. This is capturing program which prints out the headers of the packets, which transfer the network.

It is a passive software measurement tool based on the pcap-library, which includes Berkeley Packet filter. This allows running Tcpdump with different flags, which increase its functionality and allows modeling of the output display [2]. Moreover, when Tcpdump finishes capturing it reports some additional information, mainly:

- Packets received by filter – corresponds to the number of packets captured if they satisfy the particular condition.

- Packets dropped by kernel – corresponds to the number of packets dropped by the operating system, which is running on capturing computer.

In our experiments the capturing computer didn't drop any packet , so the number of packets dropped is zero.
The number of packets captured varies depending on the codecs and bit rate. The range of packets for 1024 bit rate is 81189 – 104256 packets, while for 3072 bit rate this range is 168267-271575 packets.
The size of the output pcap file is 9-11Mb for 1024 bit rate and 16-18Mb for 3072 bit rate.

The size of the movie file varies depending on the codec combination. Thus, in the following table are brought the sizes of codecs, which are detaily considered in the following section.

Codec	Bit rate	Size (kBytes)
DIVX/AC3	1024	126950
MPEG-4/MP3	1024	127031
MPEG-4/AC3	3072	215666
MJPEG/MP3	3072	340258

Table 4.1: *File size for detaily considered codecs.*

4.2.1 JITTER

For jitter, we have several values. Minimum and maximum jitter corresponds to the maximum and minimum values observed for the given interval of the movie.
Variance is a measure of how the values are spread around the expected value.

Jitter average represents the behavior of the packets in the network. It helps to give idea about the network congestion and how it manages with packets.

4.2.2 BURST

We have measured maximum burst size, maximum burst duration and burstiness. The main question for making burst duration and burst size measurements was the choice of threshold. After careful investigation we took different threshold for different bit rates. To get the adequate comparison, the threshold remained the same within the same bit rate, 10% higher than the bandwidth average bit rate for 1024 kbps bit rate, and 13% higher than bandwidth average bit rate for 3024 kbps bit rate. Burst size corresponds to the bytes stream which is greater than threshold. We took the maximum value of burst size observed for the given movie.

Burst duration represents the sum of seconds during which the byte stream was greater than threshold. Also we took the maximum value of burst duration observed for the given movie.

4.2.3 BANDWIDTH UTILIZATION

We get the average bit rate by dividing the size of the sample over its duration.

Peak defines the maximum value observed during the whole movie.

Variance defines the measure of how the values are spread around the expected value.

CHAPTER 5
COMPARISON AND ANALYSIS

In this chapter we analyze the results, plots and histograms from the previous chapter. The analysis enables us to make necessary conclusions with regards to which is the best codec for some specific use. What parameters do we have to consider? What values are more important? To begin with, we have to consider each codec combination with reference to 3 main parameters. They are jitter, burst and bandwidth utilization. Each of these parameters has certain meaning and value.

5.1 JITTER ANALYSIS

Jitter is the variation in delay of transmitted packets. It is calculated from inter-arrival time as follow:

Inter-arrival time	Jitter
T1	T2-T1
T2	T3-T2
T3	...
...	...
...	...
...	...
...	...
Tn-1	...
Tn	Tn – Tn-1

In WAN networks, jitter can have three components: physical jitter, flow control waiting time, and store-and-forward switching delay.[19] To analyze the received plots, histograms and numerical values, it is necessary to define what is desirable in terms of jitter.

Jitter can be assessed by its maximum, minimum values which define maximum and minimum possible jitter values for the given codec. It is highly desirable for the maximum and minimum jitter to have the same magnitude or be close to each other because that defines better distribution of the values.

The next parameter to be defined is jitter variance; it is defined as dispersion of the values around the mean value. Smaller jitter variance indicates less dispersion of the values. A better quality of the transmitted image is for smaller variance. The bigger values we have for variance, the more the values are spread, the more is variation in time delay, thus the probability of numerous artifacts increases.

The last parameter to consider for jitter is its average. Jitter average helps to define the packet behavior in the network in general. An average jitter close to zero is a good prerequisite for the fact that the network manages with the traffic. In our estimations we had rather ideal conditions due to small network, so we didn't deal with such issues like congestion or route change and other changes that appear in the Internet. So it gave us possibility to focus mostly on codec stream capability. And the average here was mostly the pointer on how our network was managing with particular codec.

5.2 BURST ANALYSIS

Burst is defined as a sequence of packets with size greater than a threshold t. Its main parameters are size, duration and burstiness.

Burst = sum of bytes stream > t;
Burst Size = Bytes sent in each burst;
Burst Duration = Duration of a burst;
Burst Rate = Burst Size / Burst Duration;
Burstiness = mean (Burst Duration)* mean (Gap); where mean is the arithmetic average, and gap is any period between bursts.

For the network it is common event that small percentage of flow will consume most of the network bandwidth. From burst size, we can define the peak bit rate as the maximum number of bits contained in the stream over a short predefined period of time. Burst duration defines the peak duration. The most damage to the network is brought by very high spikes of small or longer duration, so outmost attention should be paid to the burst size. We want the burst to be small in size and duration and train burstiness not long in size and in duration as well. Train burstiness is defined as a train of packets with sizes greater than threshold t. As for burstiness, it can be defined as the property of packet sequence. Burstiness should be investigated together with size and duration to clarify the picture of traffic behavior. In our analysis, primary importance was given to maximum burst duration and maximum burst size. In case these values are equal or very close in magnitude, the burstiness is a good reference to take a decision about traffic burstiness.

5.3 BANDWIDTH UTILIZATION

Bandwidth utilization is the third main important parameter in assessing the traffic. It is defined as the aggregate of all network traffic currently being transported on a path. Analysing bandwidth utilization, one can estimate what is the minimum bandwidth required for streaming the particular application. It has such properties like average bit rate, that defines how much bandwidth is required on average to stream the file. But in reality, since data are not sent continuously, but in packets, the bandwidth plot doesn't look like a straight line. Small variance makes values more predictable and that means that we can ensure the quality of service to certain degree. Peaks in plots show clear picture of how this codec behaves in relation to bandwidth utilization. We have investigated two bit rates towards bandwidth utilized and tried to find how much the increasing of bit rate affects the bandwidth utilized, whether it is proportional or there is some other dependence.

The analysis was done based on the analysis of video codec families with different audio codecs. Later, these intermediate results are compared within each other. The result is based on summary of analysis for all measurements: jitter, burst and bandwidth utilization

5.4 CODECS EXAMPLE AT 1024 kbps BIT RATE

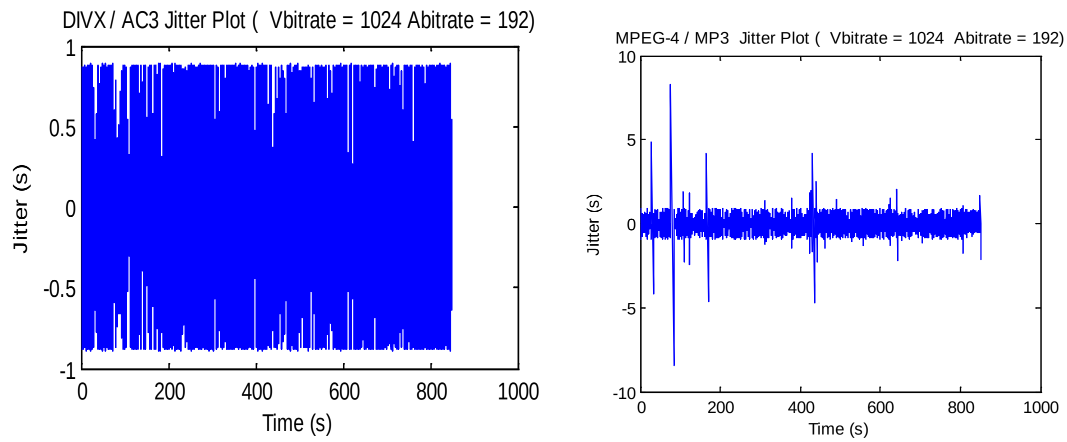
5.4.1 JITTER MEASUREMENTS

Jitter	**DIVX/AC3**	**MPEG-4/MP3**
Minimum (s)	-0.8950	-8.4590
Maximum(s)	0.8932	8.2879
Variance	0.0536	0.0256
Average(s)	-1.1793e-005	-80.127e-005

Table 5.1: *Jitter measurements for Best/Worst codecs combination at 1024 kbps.*

As seen from the experiment values in table 5.1, the jitter minimum and maximum values are smaller and more symmetric in DIVX/AC3 compared to the other codec having values more shifted. In addition several and higher spikes are observed in figure 5.2 (a) and (b) for MPEG-4/MP3. Although MPEG-4/MP3 has little smaller variance, but a significant difference in average is observed. DIVX/AC3 shows an average which is much smaller and closer to zero. This smaller average shows that the network is managing much better with packet when streaming DIVX/AC3.

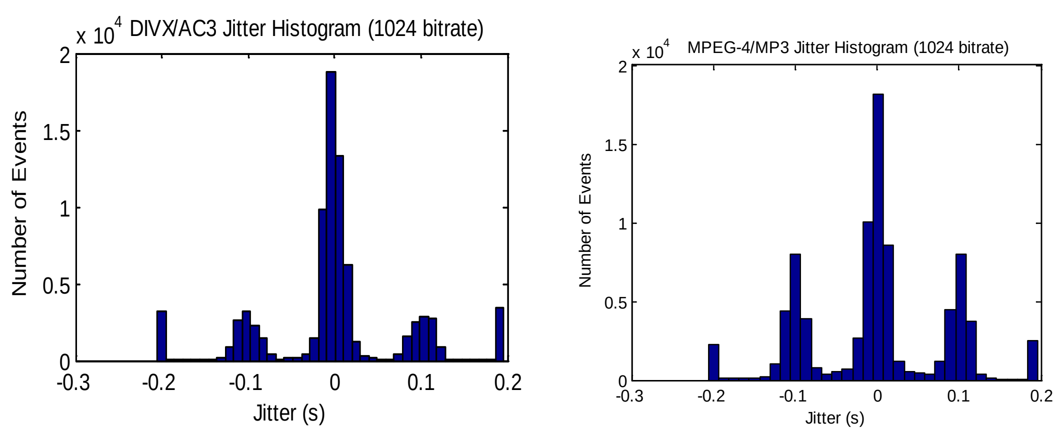
- **JITTER PLOTS FOR DIVX/AC3 AND MPEG-4/MP3**



(a) DIVX/AC3 (b) MPEG-4/MP3

Figure 5.2: *Jitter plots for DIVX/AC3 and MPEG-4/MP3 codecs at 1024 kbps bit rate.*

- **JITTER HISTOGRAMS OF DIVX/AC3 AND MPEG-4/MP3**



(a) DIVX/AC3 (b) MPEG-4/MP3

Figure 5.3: *Jitter histograms for MPEG-4/MP3 and DIVX/AC3 codecs at 1024 kbps bit rate.*

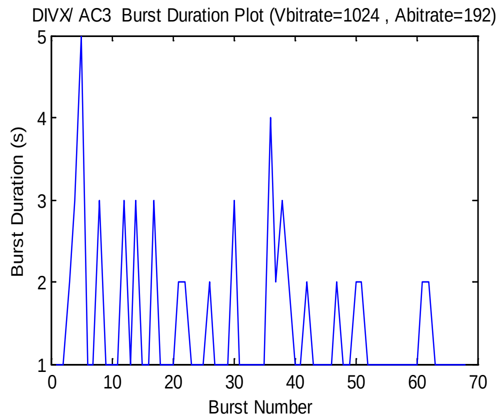
5.4.2 BURST MEASUREMENTS

Burst	**DIVX/AC3**	**MPEG-4/MP3**
Maximum Duration(s)	5	13
Maximum Size (byte)	$1.1649*10^6$	$2.7546*10^6$
Burstiness (s^2)	$3.0298*10^6$	$1.5734*10^6$

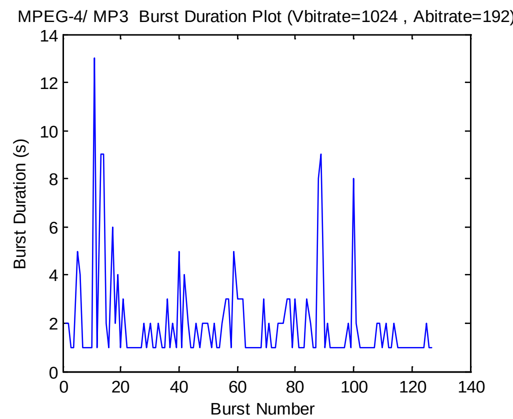
Table 5.4: *Burst measurements Best/Worst codecs combination at 1024 kbps.*

It is obviously seen from numeric values that maximum burst size and duration for MPEG-4/MP3 is roughly two times as large as for DIVX/AC3. MPEG-4/MP3 has twice as much burst than DIVX/AC3. DIVX/AC3 has greater burstiness, but MPEG-4/MP3 has greater number of burst spikes of longer duration as it is shown in figure 5.5 (b) and (d), which is more harmful for the network.

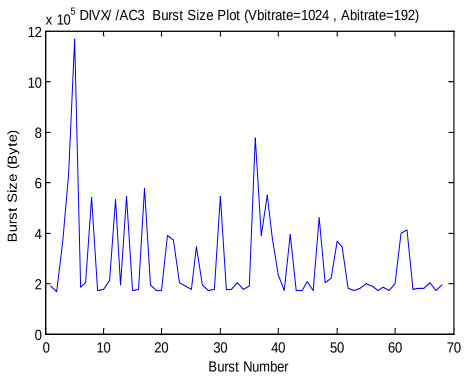
- **<u>BURST PLOTS FOR DIVX/AC3 AND MPEG-4/MP3</u>**



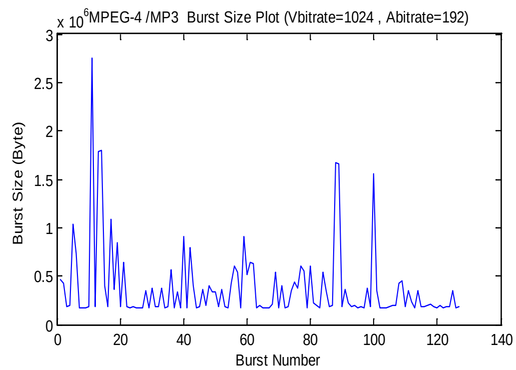
(a) DIVX/AC3 burst duration



(b) MPEG-4/MP3 burst duration



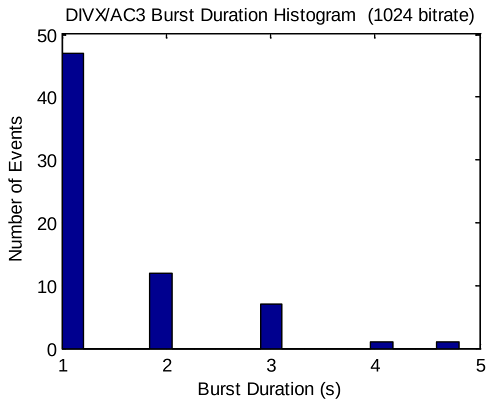
(c) DIVX/AC3 burst size



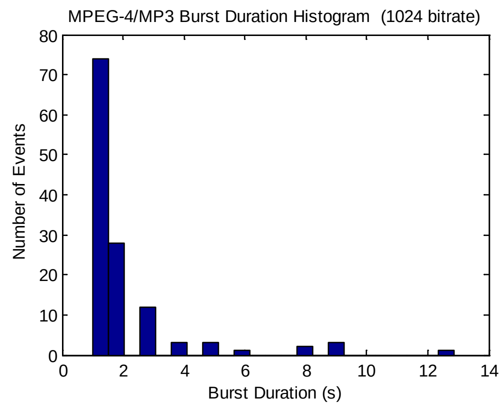
(d) MPEG-4/MP3 burst size

Figure 5.5: *Plots for Burst duration and size for DIVX/AC3 and MPEG-4/MP3 codecs at 1024 kbps bit rate.*

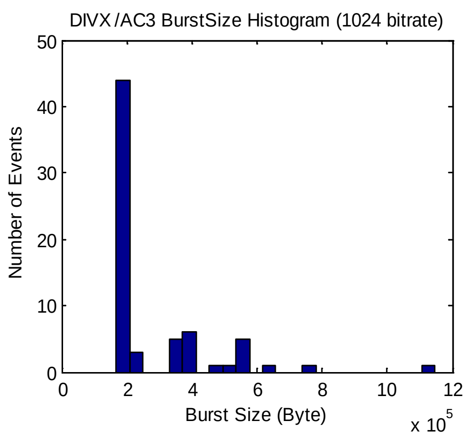
- <u>BURST HISTOGRAM FOR DIVX/AC3 AND MPEG-4/MP3</u>



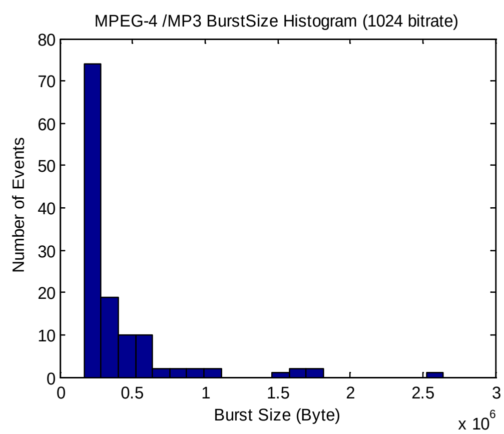
(a) DIVX/AC3 burst duration



(b) MPEG-4/MP3 burst duration



(c) DIVX/AC3 burst size



(d) MPEG-4/MP3 burst size

Figure 5.6: *Burst duration and size plots for DIVX/AC3 and MPEG-4/MP3 codecs at 1024 kbps bit rate.*

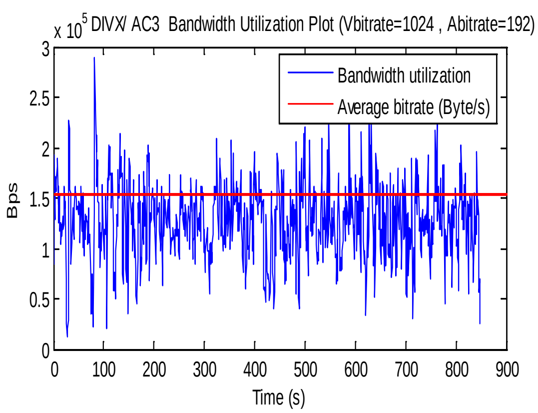
5.4.3 BANDWIDTH UTILIZATION MEASUREMENTS

Bandwidth Utilization	DIVX/AC3	MPEG-4/MP3
Average Bit Rate (Bps)	152937	153034
Peak (Bps)	289504	783520
Variance	1.4935e+009	3.0600e+009

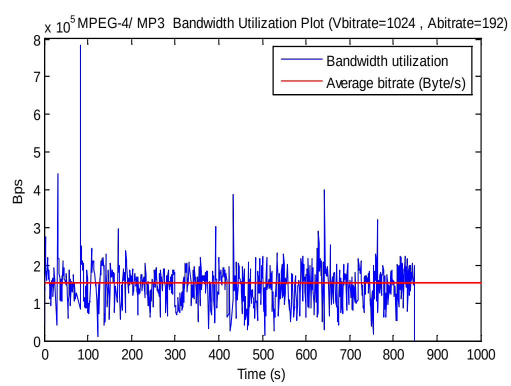
Table 5.1: *Bandwidth Utilization for Best/Worst codecs combination at 1024 kbps.*

As concerned to bandwidth utilization, we see that MPEG-4/MP3 has higher demand to minimum bandwidth required for its average bit rate. In addition to that, it has several spikes of a very high usage of bandwidth as shown in figure 5.7(b); this results in a very high variance. The high variance makes the demands for bandwidth very vague, so we can not be sure that the allocated bandwidth is enough to ensure a certain quality of service.

- **<u>BANDWIDTH PLOTS FOR DIVX/AC3 AND MPEG-4/MP3</u>**



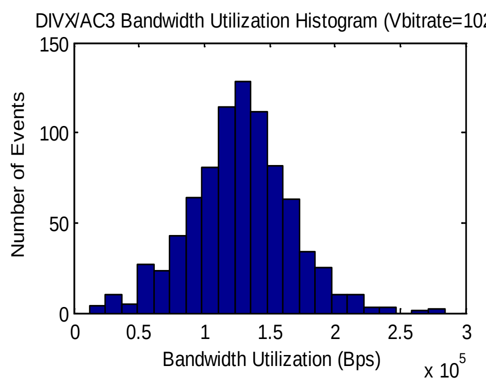
(a) DIVX/AC3



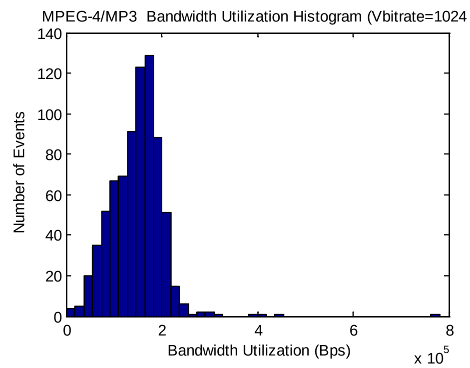
(b) MPEG-4/MP3

Figure 5.7: *Bandwidth Utilization plots for DIVX/AC3 and MPEG-4/MPEG3 codecs at 1024 kbps bit rate.*

- **<u>BANDWIDTH HISTOGRAMS FOR DIVX/AC3 AND MPEG-4/MP3</u>**



(a) DIVX/AC3



(b) MPEG-4/MP3

Figure 5.8: *Bandwidth Utilization plots for DIVX/AC3 and MPEG-4/MP3 codecs at 1024 kbps bit rate.*

5.4.4 RESULTS

Our opinion concerned these two codecs is based on the plots and measurements as well as on the visual observations. In addition to the fact that DIVX/AC3 showed better results in all statistics for jitter, burst and bandwidth, some notes about visual assessment can be done. While streaming, the MPEG-4/MP3 showed numerous artifacts like frozen screens and water effects. While for DIVX/AC3 we observed better picture quality, especially for high motion scenes. That makes DIVX/AC3 good solution for streaming, rather than MPEG-4/AC3.

5.5 CODEC EXAMPLE AT 3072 kbps BIT RATE

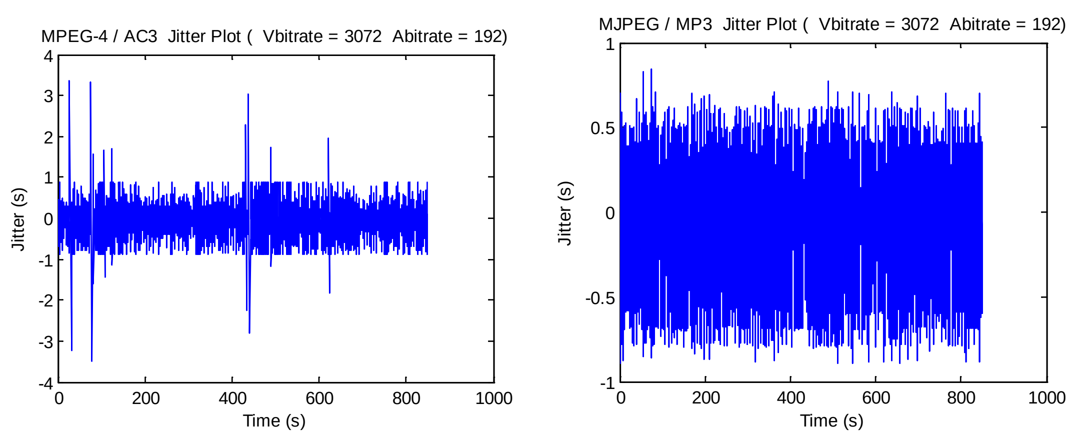
5.5.1 JITTER MEASUREMENTS

Jitter	MPEG-4/AC3	MJPEG/MP3
Minimum (s)	- 3.4904	- 0.8929
Maximum(s)	3.3634	0.8425
Variance	0.0145	0.0061
Average(s)	$-5.7701*10^{-6}$	$-6.3479*10^{-7}$

Table 5.2: Jitter measurements for Best/ Worst codec combination at 3072 kbps.

As we see, from the jitter values, MPEG-4/AC3 has worse values for maximum and minimum values, greater variance and small tail in histogram, as well as many stand alone peaks in jitter plot in figure 5.9 (a). One can conclude that MJPEG/MP3 is obviously better compared with jitter parameter.

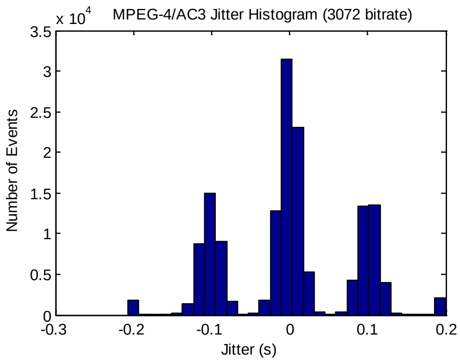
- **<u>JITTER PLOTS FOR MPEG-4/AC3 AND MJPEG/MP3</u>**



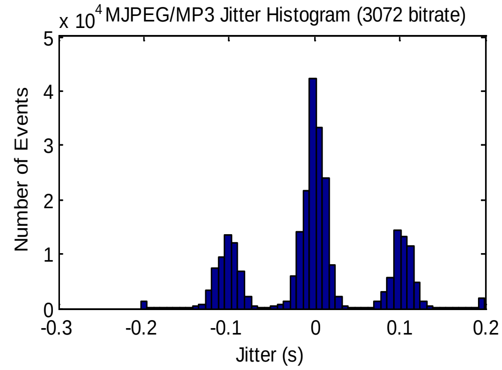
(a) MPEG-4/AC3 **(b) MJPEG/MP3**

Figure 5.9: Jitter plots for MPEG-4/AC3 and MJPEG/MP3 codecs at 3072 kbps bit rate.

- **JITTER HISTOGRAMS FOR MPEG-4/AC3 AND MJPEG/MP3**



(a) MPEG-4/AC3



(b) MJPEG/MP3

Figure 5.10: *Jitter histograms for MPEG-4/AC3 and MJPEG/MP3 codecs at 3072 kbps bit rate.*

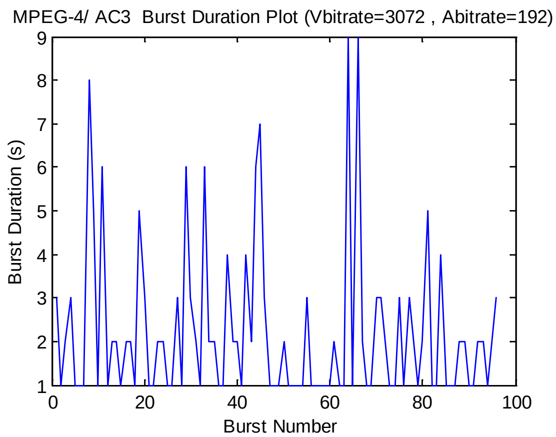
5.5.2 BURST MEASUREMENTS

Burst	**MPEG-4/AC3**	**MJPEG/MP3**
Maximum Duration(s)	9	15
Maximum Size (byte)	$3.8493*10^6$	$7.0909*10^6$
Burstiness (s^2)	$4.9182*10^6$	$16.281*10^6$

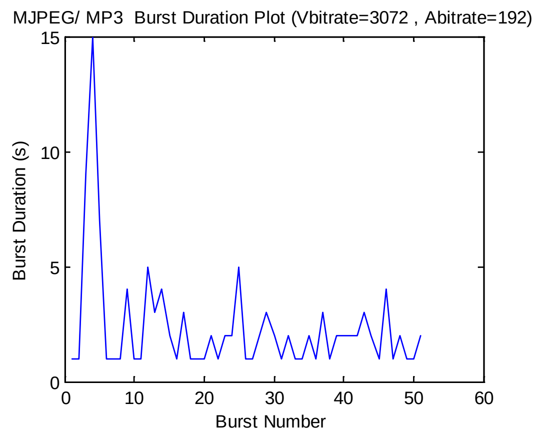
Table 5.3: *Burst measurements for Best/Worst codec combination at 3072 kbps.*

As we see, the picture is completely different for burst. For MJPEG/MP3, the maximum burst size is roughly two times as much as for MPEG-4/AC3. As concerned to the maximum duration of the burst, it is about 1.5 times greater for MJPEG/MP3 than for MPEG-4/AC3. The burstiness for MJPEG/MP3 is bigger, that defines that we have more bursty traffic during the whole interval of the movie. In addition, MJPEG/MP3 has big burst peaks with long duration that can be observed from the plots in figure 5.11 (b) and (d). As concerned to the histograms, one can say that for MPEG-4/AC3, the values are more localized and centralized, while for MJPEG/MP3 there are more dispersed and spread.

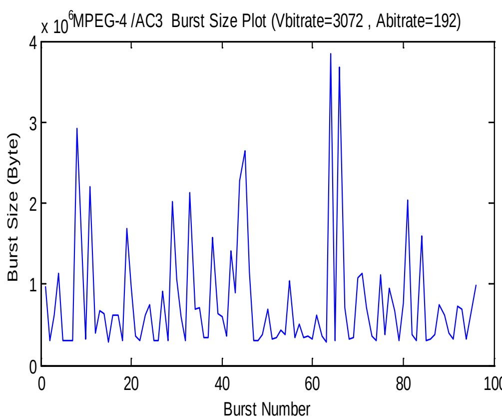
- BURST PLOTS FOR MPEG-4/AC3 AND MJPEG/MP3



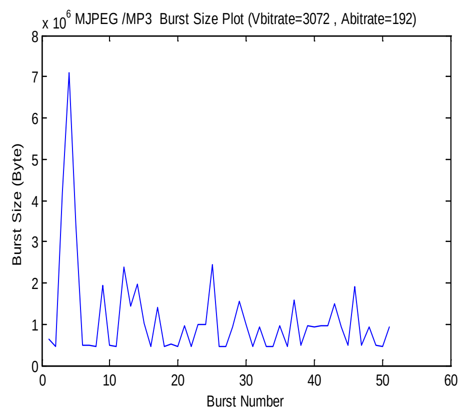
(a) MPEG-4/AC3 burst duration



(b) MJPEG/MP3 burst duration



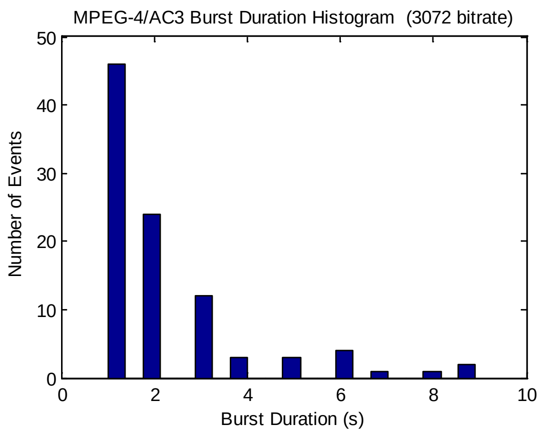
(c) MPEG-4/AC3 burst size



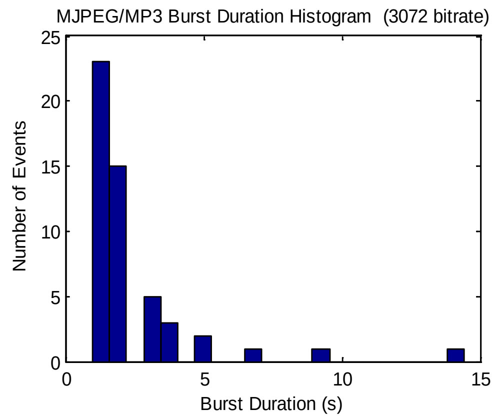
(d) MJPEG/MP3 burst size

Figure 5.11: *Burst duration and size plots for MPEG-4/AC3 and MJPEG/MP3 codecs at 3072 kbps bit rate.*

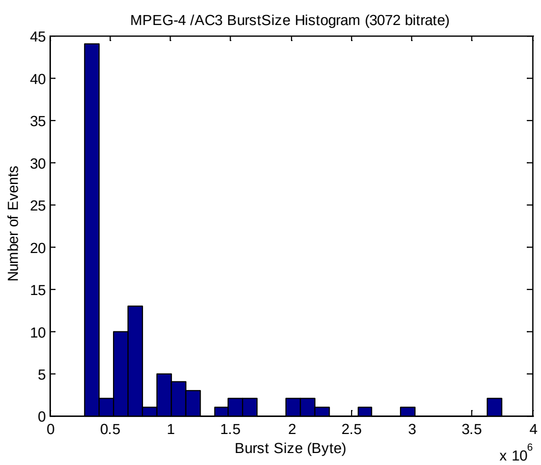
- BURST HISTOGRAMS FOR MPEG-4/AC3 AND MJPEG/MP3



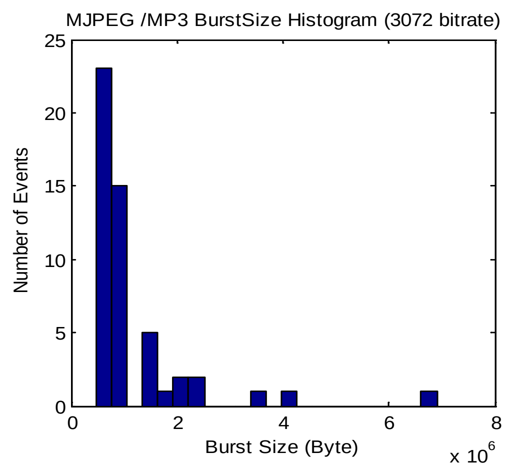
(a) MPEG-4/AC3 burst duration



(b) MJPEG/MP3 burst duration



(c) MPEG-4/AC3 burst size



(d) MJPEG/MP3 burst size

Figure 5.12: *Burst duration and size histograms for MJPEG/MP3 and MPEG-4/AC3 codecs at 3072 kbps bit rate.*

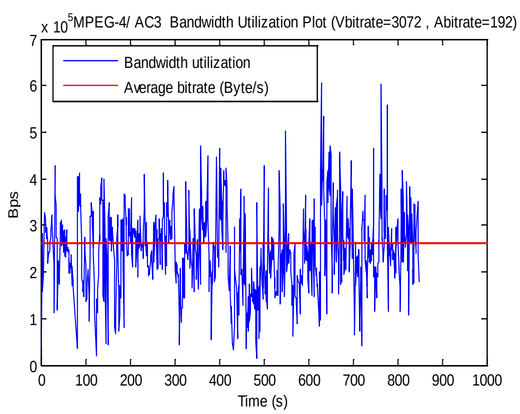
5.5.3 BANDWIDTH UTILIZATION MEASUREMENTS

Bandwidth Utilization	MPEG-4/AC3	MJPEG/MP3
Average Bit Rate (Bps)	260055	409910
Peak (Bps)	606896	650720
Variance	7.8576e+009	1.4008e+009

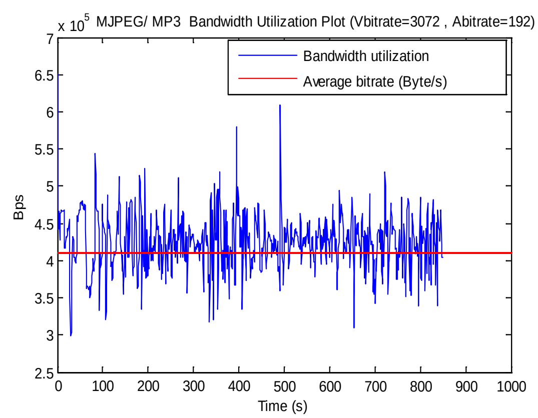
Table 5.4: Bandwidth Utilization measurements for Best/worst codec combination at 3072 kbps.

For bandwidth utilization, we see that MJPEG/MP3 requires almost two times as much as MPEG-4/AC3 bandwidth to transfer the same movie. The quality of streaming also desires better performance. Even having better parameters for variance, it couldn't be considered good at bandwidth utilization. A lot of visual artifacts, like frozen screens and dark spots, are revealed during streaming of MJPEG/MP3.
MPEG-4/AC3 bandwidth utilization is much better; it is less tailed, which means that the bandwidth is utilized better, though there are some peaks shown in figure 5.13 (b).

- **BANDWIDTH PLOTS FOR MPEG-4/AC3 AND MJPEG/AC3**



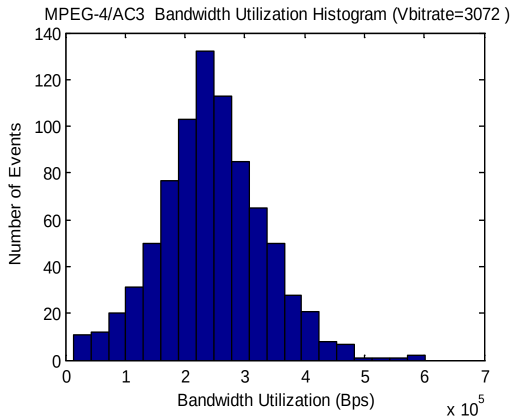
(a) MPEG-4/AC3



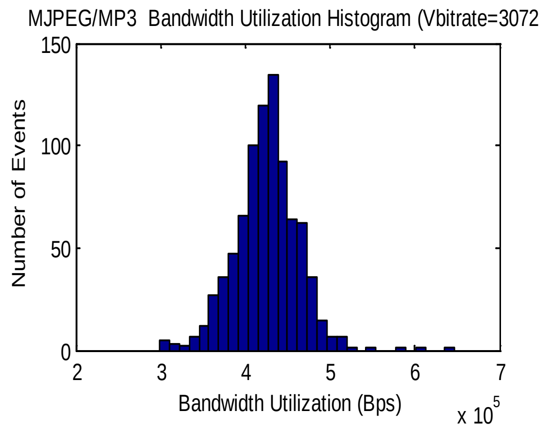
(b) MJPEG/MP3

Figure 5.13: *Bandwidth utilization plots for MPEG-4/AC3 and MJPEG/AC3 codecs at 3072 kbps bit rate.*

- **BANDWIDTH HISTOGRAMS FOR MPEG-4/AC3 AND MJPEG/MP3**



(a) MPEG-4/AC3



(b) MJPEG/MP3

Figure 5.14: Bandwidth Utilization plots for MPEG-4/AC3 and MJPEG/MP3 codecs at 3072 kbps bit rate.

5.5.4 RESULT

The results for choosing the codec are based on the jitter, burst and bandwidth measurements. The visual assessment is also taken into consideration. MJPEG/MP3 was discovered as a codec having the greatest file size from all movies considered in our thesis report. Thus, it has a highest average bit rate demand for bandwidth. A lot of visual artifacts were discovered as well. This makes MJPEG/MP3 completely unsuitable as a streaming solution.

As concerned to MPEG-4/AC3, it showed the best visual quality of picture in addition to its good statistics.

5.6 ANALYSIS AT 1024 kbps BIT RATE

5.6.1 MPEG-1 FAMILY

We have just two combinations MPEG-1/MP3 and MPEG-1/AC3.

- **JITTER MEASUREMENTS**

Jitter	MPEG-1/MP3	MPEG-1/AC3
Minimum (s)	-0.8943	- 0.8947
Maximum (s)	0.8922	0.8900
Variance	0.0514	0.0616
Average (s)	-5.5408e-006	-8.7586e-006

Table 5.5: *Jitter measurements for MPEG-1 Family.*

For MPEG-1/MP3, the jitter average value is smaller and closer to zero. However, there is no wide difference in minimum and maximum jitter values for both codec, the average for MPEG-1/MP3 is better than for MPEG-1/AC3, in addition to its smaller variance.

- **BURST MEASUREMENTS**

Burst	MPEG-1/MP3	MPEG-1/AC3
Maximum Duration (s)	10	5
Maximum Size (byte)	$1.9697*10^6$	$1.1520*10^6$
Burstiness (s^2)	$1.1157*10^6$	$3.4087*10^6$

Table 5.6: *Burst measurements for MPEG-1 Family.*

The maximum burst duration and size for MPEG-1/AC3 are less than those for MPEG-1/MP3. However we have more burstiness for MPEG-1/AC3 than for MPEG-1/MP3. MPEG-1/AC3 has better values in maximum burst duration and size.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Bandwidth utilization	MPEG-1/MP3	MPEG-1/AC3
Average bit Rate (Bps)	153039	153009
Peak (Bps)	280208	317392
Variance	7.8185e+008	8.0238e+008

Table 5.7: *Bandwidth Utilization measurements for MPEG-1Family.*

The statistics for MPEG-1/MP3 seems better because of its smaller peak and variance. It has a slightly higher average bit rate than MPEG-1/AC3, but peak is considerably smaller than that for MPEG-1/AC3. For that MPEG-1/MP3 utilizes bandwidth better than MPEG-1/AC3.

- **RESULT**

After analyzing jitter, burst and bandwidth utilization, MPEG-1/MP3 has the better result comparing with MPEG-1/AC3.

5.6.2 MPEG-2 FAMILY

We have three combinations, MPEG-2/MP3, MPEG-2/AC3, and MPEG-2/AAC.

- **JITTER MEASUREMENTS**

Jitter	MPEG-2/MP3	MPEG-2/AC3	MPEG-2/AAC
Minimum (s)	-0.8932	-0.8934	-0.8938
Maximum (s)	0.8932	0.8959	0.8924
Variance	0.0543	0.0620	0.0504
Average (s)	-5.4577e-006	-8.7930e-006	-6.3493e-006

Table 5.8: *Jitter Measurements for MPEG-2 Family.*

Analyzing jitter measurements, we came to the conclusion that MPEG-2/MP3 and MPEG-2/AAC have the best results because they have almost equal statistics. In addition, both codecs have rather symmetric minimum and maximum jitter as well

- **BURST MEASUREMENTS**

Burst	**MPEG-2/MP3**	**MPEG-2/AC3**	**MPEG-2/AAC**
Maximum Duration (s)	11	5	9
Maximum Size (byte)	$2.1241 * 10^6$	$1.1117 * 10^6$	$1.7889 * 10^6$
Burstiness (s^2)	$1.0434 * 10^6$	$3.2666 * 10^6$	$1.8931 * 10^6$

Table 5.9: *Burst measurements for MPEG-2 Family.*

Analyzing maximum burst size and burst duration values, in both statistics we can say that MPEG-2/MP3 has the biggest burst sizes with largest duration. MPEG-2/AAC has the smaller burst size and shorter duration than MPEG-2/MP3. However, MPEG-2/AC3 has the smallest maximum burst duration and smallest maximum burst size for MPEG-2 family. It can be considered as the best, even having highest burstiness value. We suppose that small burst size and duration even with more burstiness is less harmful for the network.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Bandwidth Utilization	**MPEG-2/MP3**	**MPEG-2/AC3**	**MPEG-2/AAC**
Average Bit rate (B/s)	153038	153030	161470
Peak (Bps)	312080	294816	302784
Variance	10.065e+008	7.3868e+008	7.0902e+008

Table 5.10: *Bandwidth Utilization measurements for MPEG-2 Family.*

Analyzing bandwidth utilization measurements, we noticed that MPEG-2/AC3 has the best values. Comparing the variance, MPEG-2/MP3 has very biggest variance (1.0065e+009 for MPEG-2/MP3 against 7.3868e+008 and 7.0902e+008 for MPEG-2/AC3 and MPEG-2/AAC respectively). This allows us to suggest that MPEG-2/AC3 has better bandwidth measurements.

- RESULT

After analysis and comparison, we conclude that MPEG-2/AC3 is the best combination for the considered MPEG-2 family. It shows good values for jitter, in addition to best results for burst and bandwidth utilization measurements.

5.6.3 MPEG-4 FAMILY

We have three combinations, MPEG-4/MP3, MPEG-4/AC3 and MPEG-4/AAC.

- **JITTER MEASUREMENTS**

Jitter	MPEG-4/MP3	MPEG-4/AC3	MPEG-4/AAC
Minimum (s)	-8.4590	-3.9837	-0.8924
Maximum (s)	8.2879	4.5271	0.8914
Variance	0.0256	0.0744	0.0465
Average (s)	-8.0127e-006	-1.262e-005	-6.4686e-006

Table 5.11: *Jitter measurements for MPEG-4 Family.*

We noticed that MPEG-4/MP3 is the worst between the three codecs due to its big difference in its maximum and minimum values. However even MPEG-4/AC3 has the best average, MPEG-4/AAC has better results according to all statistics in general (maximum and minimum jitter, in addition to variance).

- **BURST MEASUREMENTS**

Burst	MPEG-4/MP3	MPEG-4/AC3	MPEG-4/AAC
Maximum Duration (s)	13	5	9
Maximum Size (byte)	$2.7546 * 10^6$	$1.0958*10^6$	$1.8492 * 10^6$
Burstiness (s^2)	$1.5734 * 10^6$	$4.3610*10^6$	$1.8852 * 10^6$

Table 5.12: *Burst measurements for MPEG-4 family.*

MPEG-4/MP3 has the worst parameters among other codecs. It has longest maximum burst duration and greatest maximum burst size compared to other codecs. MPEG-4/AC3 has the best parameters due to its smaller maximum burst size and duration even having higher burstiness.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Bandwidth Utilization	**MPEG-4/MP3**	**MPEG-4/AC3**	**MPEG-4/AAC**
Average Bit Rate (Bps)	153034	153003	161618
Peak (Bps)	783520	313408	308096
Variance	3.0600e+009	2.1664e+009	7.7444e+008

Table 5.13: *Bandwidth Utilization measurements for MPEG-4 family.*

Comparing the measurements values, MPEG-4/AC3 has the best bandwidth utilization due to its small variance and average bit rate. Comparing the other two codecs, MPEG-4/AAC is the worst due to its highest bandwidth utilization and big variance.

- **RESULT**

Comparing MPEG-4 family, MPEG-4/AC3 has the best results in streaming and measurements from the other two combinations. After analyzing jitter, burst and bandwidth utilization, MPEG-4/AC3 is chosen as the best combination in MPEG-4 family for 1024 kbps bit rate.

5.6.4 DIVX FAMILY

We have three combinations, DIVX/MP3, DIVX/AC3 and DIVX/AAC**.**

- **JITTER MEASUREMENTS**

Jitter	**DIVX/MP3**	**DIVX/AC3**	**DIVX/AAC**
Minimum (s)	-0.8930	-0.8950	-0.8904
Maximum (s)	0.8913	0.8932	0.8914
Variance	0.0343	0.0536	0.0412
Average (s)	-8.7254e-006	-1.1793e-006	-7.8461e-006

Table 5.14: *Jitter measurements for DIVX family.*

DIVX/AC3 has the highest and worst variance which in addition to worst maximum and minimum jitter. DIVX/AAC has the best maximum and minimum difference, in addition to its small variance.

- **BURST MEASUREMENTS**

Burst	**DIVX/MP3**	**DIVX/AC3**	**DIVX/AAC**
Maximum Duration (s)	9	9	10
Maximum Size (byte)	$1.8744*10^6$	$1.1649*10^6$	$2.1115*10^6$
Burstiness (s^2)	$1.8543*10^6$	$3.0298*10^6$	$1.4541*10^6$

Table 5.15: *Burst measurements for DIVX family.*

Analyzing burstiness traffic, we came to the conclusion that DIVX/AC3 and DIVX/MP3 have the best values in maximum duration, maximum size and burstiness comparing DIVX/AAC. Comparing these two codecs, DIVX/MP3 and DIVX/AC3, we can note that DIVX/MP3 has greater maximum burst size, while DIVX/AC3 has highest burstiness among all codecs.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Bandwidth Utilization	**DIVX/MP3**	**DIVX/AC3**	**DIVX/AAC**
Average Bit Rate (Bps)	152990	152937	151311
Peak (Bps)	312080	289504	324032
Variance	1.4679e+009	1.4935e+009	1.0501e+009

Table 5.16: *Bandwidth Utilization measurements for DIVX family.*

Less requirements for bandwidth has DIVX/AAC due to its small average bit rate and variance. The most usage of bandwidth and big variance was observed for DIVX/MP3. Note that DIVX/AAC has greatest peak from above considered codecs. In addition, while streaming DIVX/AAC, we revealed a number of artifacts that were not discovered in DIVX/AC3.

- **RESULT**

DIVX/AC3 showed the best results in streaming and good results according to measurements in general.

5.6.5 H264 FAMILY

We have three combinations, H264/MP3, H264/AC3, and H264/AAC

- **JITTER MEASUREMENTS**

Jitter	H264/MP3	H264/AC3	H264/AAC
Minimum (s)	-0.8938	-0.8895	-0.9794
Maximum (s)	0.8916	0.8927	0.8918
Variance	0.0302	0.0319	0.0342
Average (s)	-1.9072e-005	-7.5824e-006	-8.1947e-006

Table 5.17: *Jitter measurements for H264 family.*

Analyzing jitter parameters, the best result gives H264/MP3 by all parameters minimum jitter, maximum jitter, average and small variance. The worst result shows H264/AAC. It has the worst maximum and minimum jitter, high variance and the highest average magnitude.

- **BURST MEASUREMENTS**

Burst	H264/MP3	H264/AC3	H264/AAC
Maximum Duration (s)	4	4	4
Maximum Size (byte)	$8.0345*10^5$	$8.1095*10^5$	$8.0372*10^5$
Burstiness (s^2)	$2.2339*10^6$	$2.1176*10^6$	$2.2491*10^6$

Table 5.18: *Burst measurements for H264 family*

The three codecs shown in table 5.19 have very close values in all three statistics for burst. The maximum duration is the same for all of them and small difference in maximum size and burstiness. H264/MP3 has the smallest value concerning maximum burst size while H264/AC3 has the smallest burstiness. Choosing between these two codecs, it is preferable to choose the codec having smallest maximum size which is H264/MP3.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Bandwidth Utilization	H264/MP3	H264/AC3	H264/AAC
Average Bit Rate (Bps)	161500	164336	161620
Peak (Bps)	314736	314736	339968
Variance	5.6657e+008	5.9184e+008	5.9475e+008

Table 5.19: *Bandwidth utilization measurements for H264 family.*

Comparing the codecs, we conclude that H264/MP3 and H264/AAC have the best bandwidth utilization. These two codecs have very close parameters in average bit rate, though the peak and variance are obviously higher for H264/AAC.

- **RESULT**

H264/MP3 is the chosen codec as best result between all the other considered H264 codec combination. It shows less and better usage of bandwidth and has the best statistics for jitter and burst.

5.6.6 WMV2/AAC

Only one combination of WMV2 video codec with AAC is obtained. We do not have other combination to compare it within WMV2 family. The results are brought below for later comparison with other video codecs families.

- **JITTER MEASUREMENTS**

Jitter	WMV2/AAC
Minimum (s)	-0.8820
Maximum (s)	0.8836
Variance	0.0158
Average (s)	-7.3454e-006

Table 5.20: *Jitter measurements for WMV2 family.*

- **BURST MEASUREMENTS**

Burst	**WMV2/AAC**
Maximum Duration (s)	9
Maximum Size (byte)	$1.7961*10^6$
Burstiness (s^2)	$1.9259*10^6$

Table 5.21: Burst measurements for WMV2 family.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Bandwidth Utilization	**WMV2/AAC**
Average Bit Rate (Bps)	161557
Peak (Bps)	285520
Variance	8.1881e+008

Table 5.22: *Bandwidth utilization measurements for WMV2 family.*

5.7 FINAL COMPARISON FOR 1024 VIDEO BIT RATE

- **JITTER MEASUREMENTS**

Jitter	**MPEG-1/MP3**	**MPEG-2/MP3**	**MPEG-4/AC3**	**DIVX/AC3**	**H264/MP3**	**WMV2/AAC**
Minimum (s)	-0.8943	-0.8932	-3.9837	-0.8950	-0.8938	-0.8820
Maximum (s)	0.8922	0.8932	4.5271	0.8932	0.8916	0.8836
Variance	0.0514	0.0543	0.0744	0.0536	0.0302	0.0158
Average (s)	$-5.5408*10^{-6}$	$-5.4577*10^{-6}$	$-1.262*10^{-6}$	$-1.1793*10^{-6}$	$-19.072*10^{-6}$	$-7.345*10^{-6}$

Table 5.23: *Jitter measurements for Best codec at 1024 kbps.*

Analyzing the maximum and minimum jitter for the above codecs, MPEG-4/AC3 has the worst results due to differences in magnitude for minimum and maximum values. Even this codec have small average, but it has the bigger variance compared to other codecs and very bad values for minimum and maximum which are not small, not close to zero and significantly not symmetric. We want to keep jitter closer to zero which defines better jitter, that means that our values should be as small as possible, which means less jitter and as symmetric as possible to approach to zero jitter. All other codecs, have almost symmetric values for minimum and maximum values, we have to compare other statistics like variance and average. Smaller average is observed for DIVX/AC3, but this later doesn't have good minimum, maximum and variance compared to other codecs. Variance has great importance because it defines the most undesirable values that jitter can obtain. The smaller is variance, the better is result. Here we can note two codecs that have the best values in variance: WMV2/AAC and H264/MP3.

- BURST MEASUREMENTS

Burst	MPEG-1/MP3	MPEG-2/MP3	MPEG-4/AC3	DIVX/AC3	H264/MP3	WMV2/AAC
Maximum Duration (s)	10	11	5	9	4	9
Maximum Size (byte)	$1.9697*10^6$	$2.1241*10^6$	$1.0958*10^6$	$1.1649*10^6$	$8.1095*10^5$	$1.7961*10^6$
Burstiness (s^2)	$1.1157*10^6$	$1.0434*10^6$	$4.3610*10^6$	$3.0298*10^6$	$2.1176*10^6$	$1.9259*10^6$

Table 5.24: *Burst measurements for best 1024 codec.*

For multimedia networks, it is very important to ensure the quality of service. Burst is a very undesirable event that happens unexpectedly and affects network performance. Burst peaks of great size and small duration as well as long duration negatively affect on network traffic resulting in numerous artifacts like frozen screen and black spots. So we want burst size and burst duration to be as small as possible because that is identifier of smoother traffic. We also want to have the certain consideration of burst size and duration in terms of variance and we expect it to be as small as possible. Thus, we can have predictable limits of burst. Even not having the smaller burstiness, MPEG-4/AC3 has the best values: small maximum burst size and maximum burst duration. DIVX/AC3 has good values as well.

- BANDWIDTH UTILIZATION MEASUREMENTS

Bandwidth Utilization	MPEG-1/MP3	MPEG-2/MP3	MPEG-4/AC3	DIVX/AC3	H264/MP3	WMV2/AAC
Average Bit Rate(Bps)	153039	153038	153003	152937	161500	161557
Peak(Bps)	280208	312080	313408	289504	314736	285520
Variance	$7.8185*10^8$	$10.065*10^8$	$21.664*10^8$	$14.935*10^8$	$5.9184*10^8$	$8.1881*10^8$

Table 5.25: Bandwidth utilization measurements for best 1024 bit rate codecs

In bandwidth utilization, most attention should be paid to the amount of bandwidth utilized by the codec (bit rate) in addition to its variance. The smaller bandwidth and variance are ensured and required for our codec to stream. The best results show MPEG-2/MP3 having the smallest variance, and then the second is DIVX/AC3.

- RESULT

Looking on summary and general statistics of different codecs with jitter, bandwidth and burst measurements, we concluded that DIVX/AC3 has got the best comparative results for

1024 bit rate. It has quite tolerant jitter parameters, and good burst and bandwidth parameters.
Moreover, DIVX/AC3 has excellent visual quality of streaming.

5.8 ANALYSIS AT 3072 kbps BIT RATE

5.8.1 MPEG-1 FAMILY

We have just two combinations MPEG-1/MP3 and MPEG-1/AC3.

- **JITTER MEASUREMENTS**

Jitter	MPEG-1/MP3	MPEG-1/AC3
Minimum (s)	-0.8965	-0.9753
Maximum (s)	0.8950	0.8879
Variance	0.0241	0.0157
Average (s)	-4.7698e-007	-5.0942e-006

Table 5.26: *Jitter measurements for MPEG-1 family.*

The jitter values for MPEG-1/AC3 are very bad and shifted. Jitter is measured as undesirable variation of inter-arrival time. If the delay is constant for all packets, then it is quite acceptable than it is changing its statistics. However MPEG-1/MP3 has greater variance than MPEG-1/AC3.

- **BURST MEASUREMENTS**

Burst	MPEG-1/MP3	MPEG-1/AC3
Maximum Duration(s)	179	11
Maximum Size (byte)	$1.3496*10^8$	$3.7429*10^6$
Burstiness (s^2)	$7.1632*10^6$	$3.9607*10^6$

Table 5.27: *Burst measurements for MPEG-1 family.*

For maximum burst duration and size, both values are good for MPEG-1/AC3. MPEG-1/AC3 has less maximum burst duration and less maximum burst size.
Compared to the burstiness, we have to note that here MPEG-1/AC3 is outlined by small burstiness as well.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Bandwidth Utilization	MPEG-1/MP3	MPEG-1/AC3
Average bit Rate (Bps)	252247	252242
Peak (Bps)	1332798	596272
Variance	5.1979e+010	5.8515e+009

Table 5.28: *Bandwidth utilization measurements for MPEG-1 family.*

For bandwidth utilization we have almost the same average bit rate for both codecs. Comparing other statistics, one can see that peak and variance are better for MPEG-1/AC3.

- **RESULT**

For 3072 kbps bit rate, MPEG-1 family shows better results for MPEG-1/AC3 combination.

5.8.2 MPEG-2 FAMILY

We have three combinations, MPEG-2/MP3, MPEG-2/AC3, and MPEG-2/AAC.

- **JITTER MEASUREMENTS**

Jitter	MPEG-2/MP3	MPEG-2/AC3	MPEG-2/AAC
Minimum (s)	- 0.8935	-0.8902	-0.8926
Maximum (s)	0.8904	0.8884	0.8883
Variance	0.0107	0.0127	0.0108
Average (s)	-4.2632e-006	-4.7306e-006	-3.2274e-006

Table 5.29: *Jitter measurements for MPEG-2 family.*

Analyzing jitter measurements, the best statistics for jitter has MPEG-2/MP3, for its minimum and maximum jitter value and smallest variance. MPEG-2/AC3 shows the worst result for all statistics.

- **BURST MEASUREMENTS**

Burst	**MPEG-2/MP3**	**MPEG-2/AC3**	**MPEG-2/AAC**
Maximum Duration (s)	10	9	10
Maximum Size (byte)	$4.1360*10^6$	$3.5814*10^6$	$4.1367*10^6$
Burstiness (s^2)	$3.8496*10^6$	$4.2743*10^6$	$4.2899*10^6$

Table 5.30: *Burst measurements for MPEG-2 family.*

As concerned to burst, the significant difference is observed for MPEG-2/AC3, due to its small maximum burst size and duration. MPEG-2/MP3 has smallest variance but it has great maximum burst size and duration.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Bandwidth Utilization	**MPEG-2/MP3**	**MPEG-2/AC3**	**MPEG-2/AAC**
Average Bit rate (Bps)	266727	266721	276480
Peak (Bps)	682592	581664	681264
Variance	6.5824e+009	6.5824e+009	6.5303e+009

Table 5.31: *Bandwidth utilization measurements for MPEG-2 family.*

Best bandwidth utilization parameters as MPEG-2/AC3, with significant less usage of bandwidth and smallest peaks. The greatest demand to bandwidth has MPEG-1/AAC, while MPEG-1/MP3 is outlined by highest peak.

- **RESULT**

Analyzing and comparing the results for MPEG-2 family, MPEG-2/AC3 shows best results f comparing with others. It is less bursty with small bandwidth utilization, in addition to better visual streaming results.

5.8.3 MPEG-4 Family

We have three combinations, MPEG-4/MP3, MPEG-4/AC3 and MPEG-4/AAC.

- **JITTER MEASUREMENTS**

Jitter	MPEG-4/MP3	MPEG-4/AC3	MPEG-4/AAC
Minimum (s)	-4.5170	- 3.4904	-10.6640
Maximum (s)	4.1572	3.3634	10.1597
Variance	0.0096	0.0145	0.0124
Average (s)	-4.5856e-006	-5.7701e-006	-3.3747e-006

Table 5.32: *Jitter measurements for MPEG-4 family.*

Best maximum and minimum jitter provides MPEG-4/AC3. MPEG-4/MP3 has the smallest variance. The worst minimum and maximum jitter is observed for MPEG-4/AAC. We can define that two codecs have good statistics for jitter: MPEG-4/MP3 and MPEG-4/AC3.

- **BURST MEASUREMENTS**

Burst	MPEG-4/MP3	MPEG-4/AC3	MPEG-4/AAC
Maximum Duration (s)	18	9	12
Maximum Size (byte)	$6.2380*10^6$	$3.8493*10^6$	$3.8493*10^6$
Burstiness (s^2)	$5.0989*10^6$	$4.9182*10^6$	$4.9182*10^6$

Table 5.33: *Burst measurements for MPEG-4 family.*

The worst statistics shows MPEG-4/MP3. It has the greatest maximum burst size and duration. The burstiness is the greatest for MPEG-4/MP3 as well. Comparing two other codecs according to the table 5.34, MPEG-4/AC3 and MPEG-4/AAC have equal maximum burst size and burstiness, but maximum burst duration for MPEG-4/AC3 is obviously smaller. Thus, one can consider MPEG-4/AC3 has the best result for burst statistics.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Bandwidth Utilization	MPEG-4/MP3	MPEG-4/AC3	MPEG-4/AAC
Average Bit Rate (Bps)	260059	260055	269885
Peak (Bps)	723760	606896	691888
Variance	7.8877e+009	7.8576e+009	7.2227e+009

Table 5.34: *bandwidth utilization measurements for MPEG-4 family.*

MPEG4/AC3 has the better bandwidth utilization comparing to the other codecs. It has smallest bandwidth requirements in addition to smallest peaks. The highest variance and the highest peak are observed for MPEG-1/MP3.

- **<u>RESULT</u>**

Looking through MPEG-4 family codec combinations, MPEG-4/AC3 is chosen as best result comparing with other codecs due to its good statistics for all the values: bandwidth, jitter and burst.

5.8.4 DIVX FAMILY

We have three combinations, DIVX/MP3, DIVX/AC3 and DIVX/AAC**.**

- **<u>JITTER MEASUREMENTS</u>**

Jitter	**DIVX/MP3**	**DIVX/AC3**	**DIVX/AAC**
Minimum (s)	- 0.8941	- 0.8936	- 0.8949
Maximum (s)	0.8879	0.8927	0.8849
Variance	0.0116	0.0179	0.0133
Average (s)	-4.3741e-006	-5.4072e-006	-4.2682e-006

Table 5.35: *Jitter measurements for DIVX family.*

DIVX/AC3 has the best statistics for minimum and maximum jitter, though it has the worst variance compared with other codecs. DIVX/MP3 has the smallest variance, while MPEG-4/AAC has the smallest average magnitude.

- **BURST MEASUREMENTS**

Burst	**DIVX/MP3**	**DIVX/AC3**	**DIVX/AAC**
Maximum Duration (s)	20	20	20
Maximum Size (byte)	$1.1383*10^7$	$1.0853*10^7$	$1.1396*10^7$
Burstiness (s^2)	$6.2001*10^6$	$7.8825*10^6$	$6.7335*10^6$

Table 5.36: *Burst measurements for DIVX family.*

All codecs have equal maximum burst duration, thus the conclusion should be done based on other statistics. In our analysis, we gave the primary importance for maximum duration and size. In case these values are equal or very close in values then the assessment should be done according to burstiness. DIVX/AAC has the greatest maximum burst size. DIVX/MP3 shows the smaller burstiness but its values don't differ too much compared to DIVX/AAC. DIVX/AC3 has the smaller maximum burst size, though DIVX/MP3 has the smallest burstiness. But since maximum burst size is more important value than burstiness in our decision, we choose DIVX/AC3.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Bandwidth Utilization	**DIVX/MP3**	**DIVX/AC3**	**DIVX/AAC**
Average Bit Rate (Bps)	266455	266450	276480
Peak (Bps)	925616	836640	948192
Variance	1.5685e+010	1.5724e+010	1.4708e+010

Table 5.37: *Bandwidth utilization measurements for DIVX family.*

Less bandwidth requires DIVX/AC3; it also has the smallest peaks, though not the smallest variance. Worst result shows DIVX/AAC. It has highest average bit rate and highest peak of bandwidth utilization.

- **RESULT**

DIVX/AC3 has the best results for burst and bandwidth utilization comparing to other codec combination in DIVX family. It also has good statistics for jitter that why we have chosen it as the best for DIVX family.

5.8.5 H264 FAMILY

We have one combination, H264/MP3. Thus we have no possibility to compare it within H264 family. List of measurements for jitter, burst and bandwidth utilization below will be considered for final comparison with other video codecs.

- **<u>JITTER MEASUREMENTS</u>**

Jitter	**H264/MP3**
Minimum (s)	- 0.8891
Maximum (s)	0.8430
Variance	0.0065
Average (s)	-2.7809e-006

Table 5.38: *Jitter measurements for H264 family.*

- **<u>BURST MEASUREMENTS</u>**

Burst	**H264/MP3**
Maximum Duration (s)	20
Maximum Size (byte)	$2.2073*10^6$
Burstiness (s^2)	$6.8485*10^6$

Table 5.39: *Burst measurements for H264 family.*

- **<u>BANDWIDTH UTILIZATION MEASUREMENTS</u>**

Bandwidth Utilization	**H264/MP3**
Average Bit Rate (Bps)	415052
Peak (Bps)	899056
Variance	5.1454e+009

Table 5.40: *Bandwidth utilization measurements for H264 family.*

5.8.6 WMV2/AAC

The same combination for WMV2 video codec is obtained for 3072 kbps as for 1024 kbps. This measurements will be kept for further comparison with other codec families.

- **JITTER MEASUREMENTS**

Jitter	WMV2/AAC
Minimum (s)	-0.8966
Maximum (s)	0.8908
Variance	0.0119
Average (s)	-3.8626e-006

Table 5.41: *Jitter measurements for WMV2 family*

- **BURST MEASUREMENTS**

Burst	WMV2/AAC
Maximum Duration (s)	20
Maximum Size (Byte)	$1.2113*10^7$
Burstiness (s^2)	$6.9939*10^6$

Table 5.42: *Burst measurements for WMV2 family.*

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Bandwidth Utilization	WMV2/AAC
Average Bit Rate (Bps)	309890
Peak (Bps)	1018576
Variance	1.6130e+010

Table 5.43: *Bandwidth utilization measurements for WMV2 family.*

5.8.7 MJPEG FAMILY

We have just three combinations MJPEG/MP3, MJPEG/AC3 and MJPEG/AAC.

- JITTER MEASUREMENTS

Jitter	MJPEG/MP3	MJPEG/AC3	MJPEG/AAC
Minimum (s)	- 0.8929	-0.8987	- 0.8901
Maximum (s)	0.8425	0.8891	0.8420
Variance	0.0061	0.0068	0.0057
Average (s)	- 6.3479e-007	-4.3594e-007	-1.0482e-006

Table 5.44: *Jitter measurements for MJPEG family.*

MJPEG/AC3 is outlined by its best minimum and maximum jitter among three codecs, while MJPEG/AAC has the smallest variance and average magnitude.

- BURST MEASUREMENTS

Burst	MJPEG/MP3	MJPEG/AC3	MJPEG/AAC
Maximum Duration(s)	15	3	4
Maximum Size (byte)	$7.0909*10^6$	$1.4735*10^6$	$1.4735*10^6$
Burstiness (s^2)	$1.6281*10^7$	$3.4856*10^7$	$3.4856*10^7$

Table 5.45: *Burst measurements for MJPEG family.*

From table 5.46 we can conclude that good statistics for burst have MJPEG/AC3 and MJPEG/AAC. They have equal maximum burst size and burstiness, but MJPEG/AC3 has the smallest maximum burst duration. That's why MJPEG/AC3 is considered to be better in burst values.

- BANDWIDTH UTILIZATION MEASUREMENTS

Bandwidth utilization	MJPEG/MP3	MJPEG/AC3	MJPEG/AAC
Average bit Rate (Bps)	409910	409905	421360
Peak (Bps)	650720	625488	648064
Variance	1.4008e+009	1.3506e+009	1.3325e+009

Table 5.46: *Bandwidth Utilization measurements for MJPEG family.*

MJPEG/AAC has the greatest demand to bandwidth due to its highest average bit rate. For two remaining codecs, the average bit rate is almost the same, as it is shown in the table 5.47.

Comparing MJPEG/MP3 and MJPEG/AC3 we can see that MJPEG/AC3 has smaller peak and variance than MJPEG/MP3.

- **RESULT**

From MJPEG family we have chosen MJPEG/AC3 as the best combination due to its good jitter and best burst and bandwidth utilization statistics.

5.9 FINAL COMPARISON FOR 3072 VIDEO BIT RATE

- **JITTER MEASUREMENTS**

Jitter	**MPEG-1/AC3**	**MPEG-2/AC3**	**MPEG-4/AC3**	**DIVX/AC3**	**MJPEG/AC3**	**H264/MP3**	**WMV2/AAC**
Minimum (s)	-0.9753	-0.8902	- 3.4904	- 0.8936	-0.8987	- 0.8891	-0.8966
Maximum (s)	0.8879	0.8884	3.3634	0.8927	0.8891	0.8430	0.8908
Variance	0.0157	0.0127	0.0145	0.0179	0.0068	0.0065	0.0119
Average (s)	-5.0942e-006	$-4.7306*10^6$	$-5.7701*10^6$	$-5.4072e*10^6$	$-4.3594*10^7$	$-2.7809*10^6$	$-3.8626*10^6$

Table 5.47: *Jitter measurements for Best 3072 kbps bit rate codecs.*

Analyzing the table of different codec families, we came to the conclusion that the best results has H264/MP3 for its smallest variance and average, and also good minimum and maximum magnitude, also not bad results for MJPEG/AC3 and WMV2/AAC which have small variance as well.

- **BURST MEASUREMENTS**

Burst	**MPEG-1/AC3**	**MPEG-2/AC3**	**MPEG-4/AC3**	**DIVX/AC3**	**MJPEG/AC3**	**H264/MP3**	**WMV2/AAC**
MaximumDuration (s)	11	9	9	20	3	20	20
MaximumSize (byte)	$3.7429*10^6$	$3.5814*10^6$	$3.8493*10^6$	$10.853*10^6$	$1.4735*10^6$	$2.2073*10^6$	$12.113*10^6$
Burstiness (s^2)	$3.9607*10^6$	$4.2743*10^6$	$4.9182*10^6$	$7.8825*10^6$	$3.4856*10^7$	$6.8485*10^6$	$6.9939*10^6$

Table 5.48: Burst measurements for best 3072 bit rate codecs.

Bigger maximum burst duration is observed for DIVX/AC3, H264/MP3 and WMV2/AAC. MJPEG/AC3 has the smaller maximum duration compared to others. MPEG-1/AC3, MPEG-2/AC3 and MPEG-4/AC3 show the larger maximum size. MJPEG/AC3 shows the best result having the smallest maximum duration, size and burstiness.

BANDWIDTH UTILIZATION MEASUREMENTS

Bandwidth Utilization	MPEG-1/AC3	MPEG-2/AC3	MPEG-4/AC3	DIVX/AC3	MJPEG/AC3	H264/MP3	WMV2/AAC
Average bit Rate (Bps)	252242	266721	260055	266450	409905	415052	309890
Peak (Bps)	596272	581664	606896	836640	625488	899056	1018576
Variance	$5.8515*10^9$	$6.5824*10^9$	$7.8576*10^9$	$1.5724*10^9$	$1.3506*10^9$	$5.1454*10^9$	$16.130*10^9$

Table 5.49: *Bandwidth utilization measurements for best 3072 bit rate codecs*

Comparing different codecs in the table above, MPEG-2/AC3 has the smallest peak, and quite small average bit rate and variance. MPEG-1/MP3 has average bit rate and peak values close to MPEG-2/AC3 values, but a smaller variance. MPEG-4/AC3 has also good parameters. High demands for bandwidth have MJPEG/AC3 and H264/MP3 due to their very big average bit rate. DIVX/MP3 and WMV2 have very high variance for bandwidth utilization that makes defining the bandwidth for them quite difficult.

- **RESULT**

MPEG-4/AC3 has better results for MPEG-4 family at a bit rate of 3072 kbps. It streams very well and it is less bursty comparing to other combinations.

5.10 SAME CODEC, DIFFERENT BIT RATE

5.10.1 MPEG-4/AC3

- **JITTER MEASUREMENTS**

Video Bit Rate (kbps)	1024	3072
Jitter	**MPEG-4/AC3**	**MPEG-4/AC3**
Minimum (s)	-3.9837	- 3.4904
Maximum (s)	4.5271	3.3634
Variance	0.0744	0.0145
Average (s)	-1.262e-005	-5.7701e-006

Table 5.50: *Jitter measurements for MPEG-4 codec at two different bit rates.*

Concerning jitter for MPEG-4/AC3 at 3072 kbps bit rate, we observed better minimum and maximum jitter magnitude as well. Variance is much less for 3072 kbps bit rate, which makes jitter more predictable for 3072 kbps bit rate. Though, we observed average is much higher for 3072 kbps bit rate which may create buffering problems on the client side.

- **BURST MEASUREMENTS**

Video Bit Rate (kbps)	1024	3072
Burst	**MPEG-4/AC3**	**MPEG-4/AC3**
Maximum Duration(s)	5	9
Maximum Size(Byte)	$1.0958*10^6$	$3.8493*10^6$
Burstiness (s^2)	$4.3610*10^6$	$4.9182*10^6$

Table 5.51: *Burst measurements for MPEG-4 codecs at two different bit rates.*

Observing the maximum burst size and duration, we noted that this values increase but not straightly proportionally to the increasing of bit rate. Burstiness remains almost the same for both bit rates. The considered table 5.52 shows that increment in bit rate reduce burst size or duration for MPEG-4/AC3 combination.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Video Bit Rate (kbps)	1024	3072
Bandwidth utilization	**MPEG-4/AC3**	**MPEG-4/AC3**
Average bit Rate (Bps)	153003	260055
Peak (Bps)	313408	606896
Variance	2.1664e+009	7.8576e+009

Table 5.52: *Bandwidth utilization measurements for MPEG-4 codec at two different bit rates*

Regarding the bandwidth utilization, we can note that increased bit rate doesn't result in significant improvement in bandwidth utilization. But one fact can be observed from table 5.53 that the bigger bit rate we use, the bigger is the variance for MPEG-4/AC3.

5.10.2 DIVX/MP3

- **JITTER MEASUREMENTS**

Video BitRate (kbps)	1024	3072
Jitter	**DIVX/AC3**	**DIVX/AC3**
Minimum (s)	-0.8950	- 0.8936
Maximum (s)	0.8932	0.8927
Variance	0.0536	0.0179
Average (s)	$-1.1793*10^{-6}$	$-5.4072*10^{6}$

Table 5.53: *Jitter measurements for DIVX codec at two different bit rates.*

Comparing DIVX with different bit rates, we noted that jitter remains the same with the increasing of bit rate. We can see from the table that the variance significantly decreases. This positively effects on streaming performance.

- **BURST MEASUREMENTS**

Video Bit Rate (kbps)	1024	3072
Burst	**DIVX/AC3**	**DIVX/AC3**
Maximum Duration(s)	9	20
Maximum Size(Byte)	$1.1649*10^{6}$	$1.0853*10^{7}$
Burstiness (s^2)	$3.0298*10^{6}$	$7.8825*10^{6}$

Table 5.54: *Burst measurements for DIVX codec at different bit rate.*

Regarding burst size and duration, we noted that increased bit rate results in greater size peaks and longer durations. Increasing of bit rate is increasing the burstiness as well.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Video Bit Rate (kbps)	1024	3072
Bandwidth utilization	**DIVX/AC3**	**DIVX/AC3**
Average bit Rate (Bps)	152937	266450
Peak (Bps)	289504	836640
Variance	1.4935e+009	1.5724e+010

Table 5.55: *Bandwidth Utilization measurements for DIVX codec at two different bit rates.*

Increasing the bit rate correspondingly causes the changes in bandwidth demand for DIVX/AC3. Average bit rate is increasing and the peaks are increasing as well. The variance is increased significantly that it is good prerequisite for determining the necessary bandwidth.

5.10.3 H264/MP3

- **JITTER MEASUREMENTS**

Video BitRate (kbps)	**1024**	**3072**
Jitter	**H264/MP3**	**H264/MP3**
Minimum (s)	-0.8938	- 0.8891
Maximum (s)	0.8916	0.8430
Variance	0.0302	0.0065
Average (s)	-19.072e-006	-2.7809e-006

Table 5.56: *Jitter measurements for H264 codec at two different bit rates.*

Increasing bit rate from 1024 kbps to 3072 kbps for codec H264/MP3 shifted the jitter statistics and magnitude. We observe less symmetry in the greater bit rate. The significant improvement is observed in variance. In addition, we have smaller average value for 3072 kbps bit rate.

- **BURST MEASUREMENTS**

Video Bit Rate (kbps)	**1024**	**3072**
Burst	**H264/MP3**	**H264/MP3**
Maximum Duration(s)	4	20
Maximum Size (byte)	$8.1095*10^5$	$2.2073*10^6$
Burstiness (s^2)	$2.1176*10^6$	$6.8485*10^6$

Table 5.57: *Burst measurements for H264 codec at two different bit rates.*

Increasing the bit rate caused significantly much more increment in burst size and duration for 3072 kbps bit rate. Increasing the bit rate three times results in increasing the maximum burst duration for five times and maximum size 2.5 times. Burstiness increases rather proportionally.

- **<u>BANDWIDTH UTILIZATION MEASUREMENTS</u>**

Video Bit Rate (kbps)	**1024**	**3072**
Bandwidth utilization	**H264/MP3**	**H264/MP3**
Average bit Rate (Bps)	161500	415052
Peak (Bps)	314736	899056
Variance	5.6657e+008	51.454e+008

Table 5.58: *Bandwidth Utilization measurements for H264 codec at two different bit rates.*

Bandwidth utilization looks rather proportional for both bit rates. Large increase in the bandwidth is observed which gives idea about the significant bandwidth required for the codec to stream at the higher bit rate.

5.10.4 WMV2/AAC

- **<u>JITTER MEASUREMENTS</u>**

Video BitRate (kbps)	**1024**	**3072**
Jitter	**WMV2/AAC**	**WMV2/AAC**
Minimum (s)	-0.8820	-0.8966
Maximum (s)	0.8836	0.8908
Variance	0.0158	0.0119
Average (s)	-7.3454e-006	-3.8626e-006

Table 5.59: *Jitter measurements for WMV2 codec at two different bit rates.*

After examining the measurements values for jitter, we came to the conclusion that increasing the bit rate doesn’t affect too much on the codec behavior. We observe the same maximum and minimum jitter and almost the same variance.

- **BURST MEASUREMENTS**

Video bit rate (kbps)	**1024**	**3072**
Burst	**WMV2/AAC**	**WMV2/AAC**
Maximum Duration(s)	9	20
Maximum Size (byte)	$1.7961*10^6$	$1.2113*10^7$
Burstiness (s^2)	$1.9259*10^6$	$6.9939*10^6$

Table 5.60: *Burst measurements for WMV2 codec at different bit rate.*

We can conclude that 3072 kbps bit rate has much more greater peaks of burst and longer durations. Table 5.61 shows increasing burstiness with the increment of bit rate.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Video Bit Rate (kbps)	**1024**	**3072**
Bandwidth utilization	**WMV2/AAC**	**WMV2/AAC**
Average bit Rate (Bps)	161557	309890
Peak (Bps)	285520	1018576
Variance	8.1881e+008	161.30e+008

Table 5.61: *Bandwidth Utilization measurements for WMV2 codec at different bit rate.*

As concerned to bandwidth utilization, we can conclude that average bit rate increased two times while peak is increased three times and variance is enormously increased – almost 20 times, which is highly undesirable.

5.11 SAME VIDEO CODEC, DIFFERENT AUDIO CODEC AND BITRATE

In our experiment we tried two video bit rates, 1024 kbps and 3072 kbps. Combining video codec with different audio codecs, we got a lot of similar combination which differ only in bit rate or audio codec. The following comparison is brought below for better understanding of codec behavior with increment of bit rate or increment of bit rate and changing of audio codec. The main idea was to check how the codecs evolves under this different circumstances and whether any dependences can be established.

5.11.1 MPEG-1/AC3 and MPEG-1/MP3

- **JITTER MEASUREMENTS**

Video Bit Rate (kbps)	**1024**	**3072**
Jitter	**MPEG-1/MP3**	**MPEG-1/AC3**
Minimum (s)	- 0.8943	-0.9753
Maximum (s)	0.8922	0.8879
Variance	0.0514	0.0157
Average (s)	-5.5408e-006	-5.0942e-006

Table 5.62: *Jitter measurements for MPEG-1/AC3 codec at different bit rate*

Analyzing different bit rate for the codec combinations for MPEG-1/AC3 and MPEG-1/MP3 gave us the following insight: with increasing bit rate, the magnitude for maximum and minimum is a bit shifted. Using greater bit rate for video codec, it results in less variance for MPEG-1/AC3 at 3072 kbps bit rate. The average for both codecs remains roughly the same independently of bit rate and audio codec.

- **BURST MEASUREMENTS**

Video Bit Rate (kbps)	**1024**	**3072**
Burst	**MPEG-1/MP3**	**MPEG-1/AC3**
Maximum Duration(s)	10	11
Maximum Size (byte)	$1.9697*10^6$	$3.7429*10^6$
Burstiness (s^2)	$1.1157*10^6$	$3.9607*10^6$

Table 5.63: *Burst measurements for MPEG-1/AC3 codec at different bit rate*

Increasing the bit rate for MPEG-1 video codec positively effects on burst size. It's increased only on one second, while the maximum burst size almost doubled. The burstiness also gradually increases together with bit rate.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Video Bit Rate (kbps)	**1024**	**3072**
Bandwidth utilization	**MPEG-1/MP3**	**MPEG-1/AC3**
Average bit Rate (Bps)	153039	252242
Peak (Bps)	280208	596272
Variance	7.8185e+008	58.515e+008

Table 5.64: *Bandwidth utilization measurements for MPEG-1/AC3 codec at two different bit rate.*

Concerning the bandwidth utilization, we can see from the table 5.65 the increment of average bit rate for 1.5 times, increment of peak for two times, in addition to significant increase in variance.

5.11.2 MPEG2/AC3 and MPEG-2/MP3

- **JITTER MEASUREMENTS**

Video Bit Rate (kbps)	**1024**	**3072**
Jitter	**MPEG-2/MP3**	**MPEG-2/AC3**
Minimum (s)	-0.8932	-0.8902
Maximum (s)	0.8932	0.8884
Variance	0.0543	0.0127
Average (s)	-5.4577e-006	-4.7306e-006

Table 5.65: *Jitter Measurements for MPEG-2 codec at different bit rates.*

Considering different bit rate and different audio codec for MPEG-2 video codec, we came to the conclusion that increasing in the bit rate resulted in a small shift in maximum and minimum jitter. The variance is rapidly decreasing while the average remains almost the same.

- **BURST MEASUREMENTS**

Video Bit Rate (kbps)	**1024**	**3072**
Burst	**MPEG-2/MP3**	**MPEG-2/AC3**
Maximim Duration(s)	11	9
Maximum Size(byte)	$2.1241*10^6$	$3.5814*10^6$
Burstiness (s^2)	$1.0434*10^6$	$4.2743*10^6$

Table 5.66: *Burst measurements for MPEG-2 codec at different bit rate.*

A small increment is shown for maximum burst size when increasing bit rate from 1024 kbps to 3072 kbps. The only one statistic that is increasing rather fast is burstiness.

- **BANDWIDTH UTILIZATION MEASUREMENTS**

Video Bit Rate (kbps)	**1024**	**3072**
Bandwidth utilization	**MPEG-2/MP3**	**MPEG-2/AC3**
Average bit Rate (Bps)	153038	266721
Peak (Bps)	312080	581664
Variance	1.0065e+009	6.5824e+009

Table 5.67: Bandwidth Utilization measurements for MPEG-2 Codec at two different bit rates.

From bandwidth utilization values for both codecs, it is obviously seen that greater bit rate requires more bandwidth and is characterized by greater variance as well as high peaks.

CHAPTER 6
SUMMARY OF RESULTS

In above considered work, the main criteria for choosing the appropriate codec were minimum jitter, minimum burst and minimum bandwidth utilization parameters. The conclusions about the best codec are based on these conditions. That doesn't mean that other codecs are worse or invalid. That just depends on application purposes and playback devices. They should be the main prerequisites for choosing a criterion. In particular, making accent on video codec with relation to various audio codecs and bit rate, some summary of results can be done.

6.1 MPEG-1

In our experience, there were two good combinations, MPEG-1/MP3 and MPEG-1/AC3 for both bit rates.

Comparing this two audio codecs, AC3 shows better results by all parameters, jitter, and burst and bandwidth utilization. And the same tendency continues with increment of bit rate from 1024 kbps to 3072 Kbps. Though the quality of video is low, there are a lot of visual artifacts like frozen screens. At bit rate 1024 kbps, bandwidth requirement is approximately the same like other video codec families while at increment of bit rate, demands are relatively lower than in other codecs. Perhaps the best solution for MPEG-1 is a small screen resolution displays.

6.2 MPEG-2

We have got three combinations MPEG-2/MP3, MPEG-2/AC3 and MPEG-2/AAC for both bit rates.

Among them, visually AC3 and AAC seem to be more suitable for MPEG-2 rather than MP3. Best results in bandwidth utilization and burst, though MPEG-2/AAC is better in jitter parameters and it has also demands for more bandwidth than the two other audio codecs. Certain artifacts are also discovered and frozen screen. With increase of bit rate, artifact still remains.

6.3 MPEG-4

We got three combinations MPEG-4/MP3, MPEG-4/AC3 and MPEG-4/AAC for both bit rates.

Again, MP3 shows the worst results in jitter, while MPEG-4/AAC has greater burst and bandwidth utilization, it seems that AAC audio is burstier and has increased demand in bandwidth. AC3 shows again outstanding results by its less burstiness and bandwidth. Thus AC3 can be considered as highly compatible codec because it shows the best results in combination with different video codecs. The visual quality of video is high for both bit rates and could be used as a streaming solution.

6.4 DIVX

We got three combinations DIVX/MP3, DIVX/AC3 and DIVX/AAC for both bit rates.

Jitter for all three audio codecs seems to be on the same level. Concerned to the burst, on smaller bit rate, DIVX/MP3 and DIVX/AAC show almost the same results while if the bit rate is increased, AAC shows better results. Concerned to bandwidth utilization, MP3 and AC3 have better parameters. AAC again stands for greater bandwidth. For DIVX, best audio codecs chosen were MP3 and AC3, which shows that they are both very compatible with DIVX. Visual quality is also on a high level so it could be good alternative for streaming.

6.5 H264

It should be noted that not all video/audio combinations were able to stream for chosen bit rate and we have got the following combinations:

- For 1024 Kbps bit rate we got: H264/MP3, H264/AC3 and H264/AAC combinations.
- For 3072 Kbps bit rate, we got: H264/MP3 combination only.

While streaming, the two other combinations showed damaged image.

H264 shows increasing demand for bandwidth utilization than MPEG-1 and MPEG-2 families. Its demand is similar to MPEG-4 because it is one of the profiles of MPEG-4. It seems that AC3 doesn’t match as well as MP3 and AAC since H264/AC3 combinations shows the highest burst duration and size.
Visual quality of the codec is very good.

6.6 WMV2

For this video codec, we manage to get only 1 combination which worked for both bit rates: WMV2/AAC. For other codec combinations, we got damaged video.

We couldn’t do the comparison how difference and audio codecs are mixed with WMV2. Comparing to the other video codecs, WMV2 has higher demands for bandwidth like DIVX, MPEG-4 and H264 as well as high burstiness. The visual quality is also very good.

6.7 MJPEG

Only 3072 bit rate results are available for 3 combinations: MJPEG/MP3, MJPEG/AC3 and MJPEG/AAC.

MJPEG is outlined by extremely high bandwidth requirement, as well as highest burst size and duration. In addition to that, visual image is vague and watery that makes this codec completely unsuitable for the streaming movies, especially in the case of fast action scenes.

VIDEO/ AUDIO	COMPRESSION	AUDIO support of Dolby Digital	RESOLUTION/ PROFILE	STREAMING	DRM AND LICESING VIDEO/AUDIO	APPLICATION
MPEG-1/MP3	~12 MIN	NO	Max Resolution 4095x4095/ No profiles	BAD	NO/ISSUE	Good for data storage and retrieval on place and for small resolution. Not suitable for streaming. VCD, DVD.
MPEG-1/AC3	~ 12 MIN	YES	Max Resolution 4095x4095/ No Profiles	BAD	NO/YES	
MPEG-2/MP3	~ 12 MIN	NO	YES/ Profiles-Levels.	NORMAL	YES/ISSUE	Good for data storage and retrieval, suitable for steaming as well but not so good quality as other codecs. Broadcast applications: DVB. DVD, SVCD.
MPEG-2/AC3	~12 MIN	YES		NORMAL	YES/YES	
MPEG-2/AAC	~12 MIN	NO		NORMAL	YES/YES	
MPEG-4/MP3	~20 MIN	NO	YES/ Profiles-parts.	NORMAL	LICENCED/ISSUE	Good solution for streaming, very good graphical image, but requires greater bit rates and more bandwidth. Streaming Media, video conversation, Broadcast TV.
MPEG-4/AC3	~20 MIN	YES		NORMAL	LICENCED/YES	
MPEG-4/AAC	~20 MIN	NO		NORMAL	LICENCED/YES	
DIVX/MP3	~ 8 MIN	NO	YES/Profiles	GOOD	NO FOR LATER VERSIONS/ISSUE	Good streaming solution but requires greater bandwidth. HDTV, VCD, DVD.
DIVX/AC3	~ 8 MIN	YES		GOOD	NO FOR LATER VERSIONS/YES	
DIVX/AAC	~8 MIN	NO		GOOD	NO FOR LATER VERSION/YES	
H264/MP3	~20 MIN	NO	YES/Profiles	GOOD	YES/ISSUE	Perfect streaming solution but requires high bit rate and more bandwidth. DVB,HD DVD, Blue-ray, next generation DVD
H264/AC3	~ 20 MIN	YES		GOOD	YES/YES	
H264/AAC	~ 20MIN	NO		GOOD	YES/YES	
MJPEG/MP3	~ 10MIN	NO	No/No Profiles	BAD	NO/ISSUE	Bad streaming solution and bad graphical images, not good for active movies. Mobile appliances, digital cameras.
MJPEG/AC3	~ 10MIN	YES		BAD	NO/YES	
MJPEG/AAC	~ 10MIN	NO		BAD	NO/YES	
WMV2/AAC	~ 12MIN	NO	NO/No profiles	GOOD	NO SINCE VERSION9/YES	Good solution for streaming. DVD, HD DVD.

Table 6.68: *Video/Audio Codecs Summary*

CHAPTER 7
CONCLUSION AND FUTURE WORK

As result of the work done, very broad knowledge and experience about various codecs and their behavior was gained. Different codec combinations and variations gave some curious results while applying on practice. Work was widely differentiated including different aspects - different scientific information, programming, simulation and analysis. Thesis can be considered a good manual for conducting different experiments with various codecs including encoding/decoding, streaming, sharing.

As for future work, here could be underlined the following points:

- Consideration of proper encoding/decoding and streaming application. Since real-time multimedia is relatively new service, many applications supporting it are still under the development and require proper testing and debugging.
- Further experiments with codec parameters, bit rate, resolution, etc. Realizing more experiments including strict constraints, like reducing the bandwidth available to analyze behavior of the codec under the different conditions in terms of compatibility and flexibility. That will give better representation about the codec performance and overall capability.
- Our experiment was done in ideal conditions, where the 2 computers were in the same network. This means that pure assessment of codec was done, but behavior of the codec in more complex and heterogeneous network like the Internet would include more considerations for jitter, burst and bandwidth utilization.
- Try to stream for different devices under different protocols to find out how it affects on overall performance.

All above considered points will increase the codec database for better comparison and certain recommendation for a particular codec.

*L*IST OF *F*IGURES

LIST OF TABLES

LIST OF ABBREVIATIONS

AAC	Advanced Audio Codec
AD/DA	Digital to Audio/Audio to Digital
ASF	Advanced Systems Format
AVC	Advanced video coding
AVI	Audio Video Interleave
AVO	Audio Video Objects
BIFS	Binary Format for Scene
BP	Baseline Profile
CD	Compact Disc
CIF	Common Intermediate format
DP	Data Partitioning
DVB	Digital Video Broadcasting
GOB	Group of Objects
GOP	Group of Pictures
GUI	Graphical User Interface
HDTV	High Definition TV
MP	Main Profile
MP/ML	Main Profile / Main Level
MPEG ALS	Moving Picture Expert Group-4 Audio lossless
MIME	Multipurpose Internet Mail Extensions
MUSICAM	Masking–pattern Adapted Universal Sub band Integrated Coding and Multiplexing
NTSC	National Television Standards Committee
PAL	Phase Alternating Line
PB	Predictional Bidirectional
QCIF	Quarter Common Intermediate format
RDT	Remote Digital Terminal
RS	Redundant Slices
RTP/UDP/IP	Real Time Protocol / User Datagram Protocol / Internet Protocol
RTCP	Real Time Control Protocol
SECAM	Sequentiel Couleur A Mémoire
SEI	Supplemental Enhancement Information

SIF	Source Intermediate Format
SVCD	Super Video Compact Disc
UEP	Unequal Error Protection
VCD/DVD	Video Compact Disc/Digital Versatile Disc
VCEG	Video Coding Expert Group
VCR	Video Cassette Recorder
VLC	Variable Length Coding
VUI	Video Usability Information
WAV	Waveform Audio Format
WMA	Window Media Audio
XP	Extended Profile

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