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Trading-off Compression, Energy Efficiency, and QoE In Wireless Network

Master's Thesis

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ABSTRACT

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As throughput of wireless networks has been improving rapidly over the last several years, handheld devices are becoming more and more attractive to the end users making multimedia streaming and distribution favorite applications of mobile smart devices. However, the lifetime of mobile devices highly depends on their battery power which is not growing at the pace of communication technologies. Finding a solution to minimize energy consumption while keeping the perceived quality of multimedia applications at the best possible level is of high importance. This work qualitatively and quantitatively describes the trade-off between the total energy required for communication (operation of the service, encoding and transmission), compression ratio of different audio and video codecs, and obtained quality from the network in wireless environment. The main aim of this project is to investigate the potential of implementing a system that minimizes power consumption while maintaining the best possible QoE.

We qualitatively and quantitatively describe the trade-offs between the energy required for running a service (encoding and transmission), compression ratio of different audio and video codecs, and the quality of a service as perceived by an end user in wireless environment. Having in mind a system capable to change the type of the codec on-the-fly to minimize power consumption of a device while maintaining the best possible QoE of a service running over a wire- less channel we demonstrate that power saving is, in fact, possible in both single and multiple access networks scenarios. In multiple wireless access environment these savings are not necessarily related to choosing an access network with the best throughput but dictated by the complex trade-offs between energy required for communications and propagation conditions.

In single access conditions the change of the codec should be performed more wisely compared to the common belief and depends on the choice of the parameter for optimization. Moreover, automatic modulation and coding (AMC) scheme may,

in fact, be a disadvantage for multimedia applications causing unnecessary codec changes that negatively contributes to the user perceived quality.

Results of the project will provide unified solution for joint performance and energy optimization for media applications running over wireless channels. A real-time system responsible for dynamic optimization of energy and QoE metrics can be developed in future studies.

PREFACE

This project is the final proof of Master of Science (MSc) Thesis, Trading-off Compression, Energy Efficiency, and QoE In Wireless Network, in the Department of Communications Engineering at Tampere University of Technology, Finland. The executed research has been done during the year 2012-2013 at the Institute of Communication Engineering, Tampere University of Technology.

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CONTENT

ABBREVIATIONS AND DEFINITIONS

QoE	Quality of Experience
WSN	Wireless Sensor Network
4G	4th Generation
IP	Internet Protocol
ARQ	Automatic Repeat-reQuest
FEC	Forward Error Correction
PDU	Protocol Data Unit
MIMO	Multiple-Input and Multiple-Output
SNR	Signal to Noise Ratio
ICT	Information and Communications Technology
QoS	Quality of service
ITU	International Telecommunication Union
VoIP	Voice over IP
RLE	Run Length Encoding
VQ	Vector Quantization
DCT	Discrete Cosine Transform
DWT	Discrete Wavelet Transform
PSNR	Peak Signal to Noise Ratio
SSIM	Structural SIMilarity
MOS	Mean Opinion Score
PLR	Packet Loss Ratio
TVI	Time Varying Impairments
RE	Recency Effect
VQA	Video Quality Assessment

MSE	Mean Square Error
HVS	Human Visual System
MIPS	Million Instructions Per Second
EC	Error Concealment
DSP	Digital Signal Processor
CS-ACELP	Conjugate Structure Algebraic Code-Excited Linear Prediction
ADPCM	Adaptive Differential Pulse Code Modulation
QCIF	Quarter Common Intermediate Format
SISO	Single Input Single Output
LTE	Long-Term Evolution
TTI	Transmission Time Interval
UMTS	Universal Mobile Telecommunication System
PSTN	Public Switched Telephone Network
SIP	Session Initiation Protocol
BER	Bit Error Rate
DoS	Denial of Service
IETF	Internet Engineering Task Force
SDP	Session Description Protocol
TCP	Transmission Control Protocol
PPS	Packet Per Second
MDCT	Modified Discrete Cosine Transform
BWE	BandWidth Extension
AVQ	Algebraic Vector Quantization
FLVQ	Fuzzy Learning Vector Quantization
TD	Time Differential

FD	Frequency Differential
MLT	Modulated Lapped Transform
TV	Time Variant
cRTP	compressed Real-time Transport Protocol
MLP	Multilink Point-to-Point Protocol
FRF	Frame Relay Forum
FCS	Frame Check Sequence
CRC	Cyclic Redundancy Check
PSQM	Perceptual Speech Quality Measure
PESQ	Perceptual Evaluation of Speech Quality
PAMS	Perceptual Analysis Measurement System
MOPS	Million Operations Per Second
WMOPS	Weighted Million Operations Per Second
ETSI	European Telecommunications Standardization Institute
OFDMA	Orthogonal Frequency Division Multiple Access
W-CDMA	Wide-band Code Division Multiple Access
RTP	Real-time Transport Protocol
RTCP	RTP Control Protocol
ASF	Advanced Systems Format
AVI	Audio Video Interleave
ISO	Internal Standards Organization
CAVLC	Context-Adaptive Variable-Length Coding
CABAC	Context-Adaptive Binary Arithmetic Coding
SVC	Scalable Video Coding
MVC	Multi-view Video Coding

SBP	Scalable Baseline Profile
SHP	Scalable High Profile
CP	Complexity Parameter
QP	Quantization Parameter

1. INTRODUCTION

Over the last several years we have witnessed a tremendous growth of mobile market. From one side it was stimulated by ever increasing throughout provided by wireless access network nowadays reaching 100Mbps downlink for cellular systems and 500Mbps for local wireless access. Evolution of mobile computing over these years was on pace with wireless technologies providing us with small and mid-size hand-held mobile equipment capable of storing large arrays of information and performing complex processing tasks. Coupled with with the current state of networking media services these two factors enabled penetration of these services at the air interface finally starting to fulfill the long-standing dream of networking community by providing any currently available service anytime, anywhere.

One of the problems greatly affecting user experience of modern hand-held devices is their extremely limited lifetime. The main reason is usage of advanced functional components such as high-definition displays, multi-core central and graphical processing units, complex modern wireless interfaces, etc. The evolution of these components happen at rather fast pace. As an example, over the last decade we jumped from simple monochrome screen to multi-touch OLED (organic light-emitting diode) displays, from "simple" 4377069 microcontroller working at 1Mhz in the most popular ever Nokia 1100 to multi-core CPUs in modern smartphones and tablets, from no dedicated GPU to multi-core GPUs nowadays.

All in all, we may fairly claim that we experienced a qualitative improvement in performance of mobile components1. At the same time over this timespan there were no significant breakthroughs in battery technologies [1] , [2]. In fact, those hand-held devices produced in the beginning of 2000s were already started to be equipped with lithium-ion (Li-Ion) batteries providing lifetime of up to few weeks in stand- by regime and up to few dozens of hours of talk time. These batteries with slightly improved capacity are still used nowadays. Improving their capacity is hardly feasible as this is only possible via increasing the size and/or the number of working elements. Conserving energy to extend the lifetime of modern mobile devices is thus an important problem.

It is important to note that modern smart phones very frequently provide longer lifetime in both stand-by regime. The reason is quantitatively increased battery life and various sleeping regimes used in modern phones. Once a smart-phone is not used the display is turned off after some time, then, CPU goes to the sleeping mode. During the sleeping mode network interfaces maintained at their minimum functional capability, e.g. listening over control channels in cellular systems. For details one may refer to [3]. Note that even in talk regime all smart-phones turn off their displays spending energy for running the conventional voice call service only.

The difference nowadays is when a smart-phone is in-use. The set of services provided by modern hand-held devices became exceptionally rich over the last few years with introduction of current generation of cellular systems and embedded support for IEEE-based wireless local area networks (WLAN). In fact, all those multimedia applications available with wired connection became accessible for mobile users. These services consume a lot more energy compared to conventional cellular telephony due to usage of complex media encoding schemes. At the same time modern wireless access technologies become more and more complicated involving sophisticated channel adaptation mechanisms and, as a result, requiring more energy for communication.

In this thesis we do not consider energy consumption of mobile devices during their whole lifetime. Instead, we address the way how energy can be optimized when running modern media services such as voice-over-IP (VoIP) and video delivery (streaming, conferencing, on-demand). We are also not concerned with protocol modifications of modern wireless access technologies that may potentially lead to energy savings. Instead, we concentrate on those factors contributing to energy requirements of applications running on mobile devices including energy required for encoding and energy required for communication over the air interface. However, this question cannot be nowadays considered independently of other issues pertaining to quality of provided service. In fact, we should also take into account other performance issues related to providing services over the wireless channels.

Performance of a networking services are conventionally estimated using the notion of quality of service (QoS) introduced in the beginning of 90s. Aside from service support performance metrics, e.g. service maintenance and support, service availability, that are not directly related to the network performance the set of specifications issued by ITU-T includes metrics related to traffic performance such as IP packet loss ration (IPLR) and IP packet transfer delay (IPTD). These metrics were specified for several media applications including VoIP and video conferencing and were then deemed to correctly describe the service quality as experienced by the end user.

Since then, in spite of tremendous growth of media applications, these metrics have not been updated. There are a number of reason behind it. First of all, these metrics were found to produce rather large deviations compared to conventional quality assessment techniques for video, speech and audio. Secondly, there are no analytical methods to convert them to more convenient and understandable scales (e.g. bad, fair, good) that are usually used describing the so-called perceived quality. For this mapping to be possible one had to get rather large statistical database using conventional tests based on mean opinion score (MOS) scale. What is really important is that such a study is required for each particular media codec. Yet another obstacle towards description of perceived quality based on traffic-related QoS metrics is their complex behavior.

Indeed, both network delay and loss metrics are in fact stochastic processes that may change their local behavior in time. Averaging their values over time may not be a proper way to describe performance of media applications. The reasons vary from different importance of data in encoded bitstream to complex psychovisual effects affecting the way impairments are perceived. Finally, degradation of perceived quality not only happens inside the network but introduced by codecs themselves. Abundance of codecs available for encoding and compression of media information in the beginning of 2000s made standardization efforts of network-based metrics rather obsolete. The problem of assessing quality of media applications has also been approached by researcher working with encoding and compression of media information. The main shortcoming of such metrics is that they cannot be implemented in network environment as most of those require original media at the evaluation point.

In wireless networks performance degradation of applications in wireless- cum-wired usually stems from rather limited resources (e.g. limited bandwidth of a shared or dedicated channel) and error-prone nature of a wireless channel caused by dynamically changing wireless channel characteristics. The way how the latter manifests itself highly depends on a given wireless access technology, however, in most cases a system similar to automatic modulation and coding (AMC) is used to maintain performance of a channel at some acceptable level. As a metric of interest, packet loss probability is used as an ultimate measure of performance. AMC systems allows to apply different modulation schemes and forward error correction codes (FEC) such that this metric is optimized.

Special channel organization techniques such as orthogonal frequency-division multiplexing (OFDM) as well as bit or symbol interleaving techniques adds to "randomization" of a wireless channel producing better environment for AMC to work with. On top of AMC system a certain type of automatic repeat request (ARQ) mechanism is sometimes used to deal with those incorrectly received blocks/frames left after AMC procedure.

Depending on the selected parameters of AMC and ARQ mechanisms a certain amount of corrupted data still propagates to higher layers resulting in loss of IP packets. Since RTP/UDP protocols are used at the transport layer incorrectly received packets are not recovered and eventually affects perceived quality of media applications. Depending on the type of the codec currently used.

Modern media applications nowadays include possibility of choosing the type of the codec their parameters (if applicable). Furthermore, some modern programs even allows to change parameters of a chosen codec on-the-fly. This is done in those programs implementing real-time video services to adapt the rate of the codec to that currently provided by the network. From our point of view capabilities of this system can be significantly extended to not only provide rate adaptation to changing network conditions but to optimize the perceived quality experienced by the user while maintaining energy consumption of the service at the minimum possible level in both single and multi-access environments.

For this reason we investigate trade-offs and dependencies between perceived quality of media applications, the amount of energy required to run the service including encoding/compression operations and communication energy overhead for different wireless access technologies and different packet loss probabilities. For both VoIP and video services we provide both qualitative and quantitative results. Our main qualitative conclusions are (i) depending on the type of wireless access technology and the type of a codec the amount of energy spent for communication can be bigger or smaller compared to the amount of energy spent for encoding potentially allowing for joint energy/quality optimization, (ii) media applications may tolerate significant changes in the packet loss rate without significant changes in the perceived quality.

All these taken together imply that a system simultaneously optimizing both QoE and energy consumption of media application may actually be beneficial for mobile devices extending their lifetime. In some cases, especially, in favorable multi-access environment these saving can be significant. In some other case the gain can be insignificant. However, taking into account that the system adapting the rate provided by the network us currently implemented anyway in many media applications it would be wise to couple it with QoE and energy optimization features.

1.1 Motivation

With the commercial launch of 4th Generation (4G) mobile systems, the convergence of the Internet and mobile communications is becoming clear. This process is stimulated by the rapid growth of both technologies, as well as by users who wish to readily access Internet services. In addition to providing broadband wireless access to the Internet, modern and future wireless networks are expected to satisfy QoE requirements of various media applications. This is an inherent problem for many service types, even in fixed Internet Protocol (IP) networks. Wireless and mobility issues add their own problems on top of this IP flaw.

The time-varying erroneous nature of wireless channels, cross-layer organization of the protocol stack, and tele-traffic and mobility issues shoud be addressed before wireless media Internet services can be widely deployed. The intention to adopt IP protocol for future mobile communications and the extension of Internet services to the air interface necessitates novel performance evaluation and optimization methods.

Since the error rate of a fixed transmission medium is negligibly low, to evaluate the performance of applications in fixed IP networks, it is sufficient to estimate performance degradation caused by packet forwarding procedures. When dealing with wireless networks, we must also take into account performance degradation caused by bit errors of wireless channels. This degradation, which significantly contributes to end-to-end performance expectations, has a completely different nature than that seen in fixed networks. The air interface is expected to be a weak point in any QoE assurance model.

To provide optimized performance, wireless technologies implement a number of channel adaptation mechanisms. These are Automatic Repeat-reQuest (ARQ) and Forward Error Correction (FEC) techniques, adaptive size of Protocol Data Unit (PDU)s at different layers, Multiple-Input and Multiple-Output (MIMO) antenna design, power control mechanisms, etc. All of these mechanisms are implemented at different layers of the protocol stack and affect performance provided to applications differently. To provide their unified and optimized operation, wireless technologies call for novel design of the protocol stack that should include cross-layer performance optimization capabilities.

The recent improvement in processing and storage capabilities of mobile devices makes them suitable for computationally media intensive services, e.g., video streaming. Such services are characterized by strict QoE metrics, but add another complicating factor related to excessive energy consumption of mobile devices. The energy should be spent not only on transmission of actual data, but for encoding/decoding operations. Given the dearth of breakthroughs, in battery technology, this problem will only get worse as more capable devices and services hit the market. What is more important is that the problem of providing satisfactory QoE is inherently related to the energy consumption of various transmission technologies, and this relationship is not linear.

Theoretical and practical results of the project will provide basic ideas of a unified solution for joint performance and energy optimization for media applications running over wireless channels. A real-time system responsible for dynamic optimization of energy and QoE metrics can be developed. The performance control and optimization system, if properly formulated, designed, and implemented, can be of high practical importance.

Focusing on energy considerations makes it possible to design a potential system that may trade-off energy for air interface with one factor used for compression and coding. A QoE-aware system should use the most appropriate codec (compression algorithm) to optimize power consumption and ensure a desired level of quality.

This total energy is the addition of transmission and compression and coding. Each coding method results in a certain level of QoE in terms of a specific level of energy. Defining a threshold for both QoE and energy expenses allows the potential system to modify its transmission technology or complexity of codec in order to make the best choice that satisfies all limitations.

1.2 Energy-efficient communication

In the context of energy consumption in wireless networks, the effect of communication equipment cannot be neglected as a major energy consuming component [4]. Two main factors to be modified in order to reduce energy consumption are transmission time and transmission power. Increasing transmission bitrate results in lower transmission time but requires more transmission power to satisfy Signal to Noise Ratio (SNR) constraints. This means that transmission time and power are related, and trade-offs between these two must be examined. [5] investigated both theoretical formulas given by Shannon and their own method for energy consumption in wireless networks.

Considering the previously mentioned power consumption issues, there are two major solutions to remedy the short lifetime of portable devices.The first is to improve the design and capacity of batteries. However, these devices are small and lightweight; as a result, designing batteries is a challenging task.

It is important to mention that the capacity of batteries has improved in recent years, but not at the same pace of Information and Communications Technology (ICT) industry and wireless revolutions[6], [1]. In fact, new batteries are mainly used on higher power consuming devices to maintain runtime of device at the same level of old, less energy consuming devices, at the best case scenario. This deficiency in their capability has made it necessary for researchers to find and introduce energy efficient algorithms in wireless networks; this is the second solution for increasing the lifetime of new powerful portable devices.

Various protocol optimization schemes might enable power management techniques during transmission. The method introduced in [7] focuses on data link layer and discusses retransmissions in an energy constraint environment.[8] focused on multimedia streaming and the effect of traffic structure within network and radio access technology used for transmission.

In addition to improving the lifetime of portable devices and their performance, there are other important factors motivating researchers to work on energy efficiency in communication networks. One such factor concerns global environment issues. According to [9], 15-20 percent of the entire communications industry and 0.3-0.4 percent of annual carbon dioxide emissions relate to mobile communications.

Although there have been some studies done in this field, none have compared all three metrics of QoE, compression, and energy efficiency These are the areas addressed in this thesis.

1.3 Modern media applications and QoE

For a long time, Quality of Service (QoS) was the basic concept used by researchers to show the performance of applications. However, it was necessary to use a metric to provide a better understanding of a user's perception. QoE, the currently used metric, shows user requirements and level of satisfaction with a specific application.

The concept of QoS defined by International Telecommunication Union (ITU) is the totality of characteristics of a telecommunications service that bear on its ability to satisfy the stated and implied needs of the user of the service (ITU-T Rec. E.800 (2008)). On the other hand, QoE can be defined as how satisfied a user is after using an application.

QoE evaluations are conducted using two different approaches. The first approach, called subjective testing, is based on user perceptions and is usually used in laboratories. The second approach, which is more objective, is based on instrumental calculations such as E-model in speech communication applications. Subjective tests are time-consuming and expensive; thus, objective tests are favorite solutions to calculate QoE value of an application or service. In addition, a well defined QoE model make online monitoring of quality possible. However, this model must not be too complex to be applicable in real-time applications.

In multimedia applications, one of the most important factor affecting quality is the compression algorithm used in codecs for digitizing and compressing audio and video. Codecs used in multimedia applications mostly use lossy compression techniques; however, the way in which they treat signals can vary. More advanced codecs use better compression schemes by means of implementing advanced methods resulting in higher complexity.

Complexity of codecs is one of the most important factors to consider when choosing a codec for a specific application. Higher complexity (more advanced codecs) might result in a better QoE and compression ratio, which means less energy is required for transmission. However, the increasing complexity of codecs means more energy required for processing a compression algorithm.

In addition to processing power, another effect of complex codecs that use advanced compression techniques is related to the effect of packet losses on transmitted data. In applications that use these codecs, a huge amount of data would be compressed into packets and be transmitted over the channel. In this scenario, loosing a single packet causes more degradation in quality than in codecs with lower compression ratios. In addition to this degradation, losing some consecutive packets might even cause the application to drop the session.

For applications in which video data must be compressed and transmitted, more complex techniques result in a noticeable amount of power consumption because of the inherent characteristics of video contents. However, this issue also exists in the case of speech codecs used in Voice over IP (VoIP) applications, where a more complex codec would require much more energy than simpler codecs.

Describing video compression schemes in details requires dense mathematical equations. However, in order to have a better understanding of why complexity in video codecs is significant , one must understand different technologies and compression algorithms:

1. The most primitive algorithm, Run Length Encoding (RLE), is used for encoding consecutive pixels of the same color. The encoded output would be a single codeword. However, RLE cannot be used widely, as it is not common to have runs of the same color. An example of RLE is encoding the uncompressed sequence "AAAABBBCCDDDDDD," which requires 13 Bytes of storage space; after encoding by RLE, it becomes "4A3B2C4D" (4 of As, 3 of Bs, 2 of Cs, and 6 od Ds). In this example, the compression ratio would be almost 2 to 1.

2. The basic idea of Vector Quantization (VQ) to divide an image into blocks. In this algorithm, different blocks of an image are compared; similar blocks are gathered into one class. Each of these classes is presented using a code in binary. Then, the encoder simply inputs one generic block instead of all of the original blocks of each class. It is then only necessary to encode the binary code of each generic block using the lookup table that maps these blocks to their corresponding binary codes.

   In order to improve the efficiency of codec, more frequently occurring generic blocks are presented using shorter binary codes. At the other side, the decoder uses the lookup table to reassemble generic blocks from the binary codes. This algorithm is a quick, lossy compression algorithm.

3. Discrete Cosine Transform (DCT) is widely used in different codecs such as MPEG-4 and H.263 video conferencing standard. In this technique, a two-dimensional DCT is applied to 8*8 blocks of pixels. Then the resulting coefficients are quantized by a quantization factor. In the next step, RLE is applied to these quantized coefficients. Most DCT coefficients are small; they become

zero after the quantization process. Such factors are considered because the human eye is less sensitive to higher frequencies; thus, a large quantization factor is applied to higher frequency components.

4. The concept Discrete Wavelet Transform (DWT) involves passing the original signal through a low pass filter to obtain lowered resolution of the signal. The original signal is also passed through a high pass filter to obtain a detailed version of the signal. Using downsampling by two, the addition of these two signals does not affect the original number of bits. Choosing appropriate parameters for filters, upsampled versions of two signals results in the original signal. Up to this step, there is no compression, as the original signal with the same number of bits has been regenerated. In order for compression to happen, VQ is applied to the detailed signal coefficients. After this quantization, some coding techniques are also applied to the quantized coefficients; one of these techniques is entropy coding. Based on the quantization factor and the coding algorithm applied to quantized coefficients, the compression ratio can be managed.

5. Motion Compensation is used for coding motions within a scene such as a moving train. In fact, this algorithm is a combination of a number of different algorithms. For compensating purposes,the method chooses a reference picture; this might be a previous or future picture. The method then describes an image based on that reference picture and the current one. In other words, it uses the fact that in many frames of a movie, most of the objects remain stationary. As a result, it is only necessary to store the information needed to present these transformations.

All of these techniques and those used in speech codecs require different levels of complexity and amounts of energy. They might improve QoE after compression because of more advanced techniques; at the same time, such methods might degrade QoE after transmission due to lost or compressed information. Higher complexity, which mostly means higher compression ratios, requires less transmission energy, but more processing power. That is why choosing the best possible codec depends on many factors, such as network condition, application requirements, and battery capabilities; it is not a straightforward task.

1.4 Structure of the thesis

This thesis is organized as follows:

Chapter 2, explains each of three components that this report presents in detail. The first section of Chapter Two describes the reason for compression. Section two describes QoE metrics in both speech and video codecs, and the sources of degradation of QoE are introduced and examined. The last section presents all definitions of energy requirements and compares various compression and transmission power consumption schemes.

Chapter 3, on speech codecs, begins with existing challenges in VoIP. Then different standards and protocols used in VoIP applications are introduced to provide a better overview of these services. Different codecs are compared in terms of the compression algorithms used resulting in various QoE values, processing power, and processing power consumption. Finally, the energy required for transmission over different radio access technologies for each codec is presented and compared to corresponding processing power. The terms "audio" and "speech" are used interchangeably in this thesis.

Chapter 4, video codecs, defines the basic requirements and necessities for communications containing video. The widely used video codec H.264 and the structure of its encoder and decoder are then described briefly. Different profiles and levels of H.264 are suitable for various applications; those most suitable for real-time applications are discussed. Finally, the proposed method for trade-offs in H.264 and the resulting graphs and tables are presented.

Chapter 5, the conclusion, proves that there is a potential for trading-off QoE, compression, and energy consumption that is required when implementing a system that can choose the best codec in different conditions.

2. DESCRIPTION OF TRADE-OFFS

For a long time quality of service (QoS) was the concept used by researchers to describe the performance of networking applications. These metrics rely on network-intrinsic parameters such as packet loss ration and mean packet delay and do not provide simple relations to conventional scale. Quality of user experience (QoE), the currently used metric, shows user requirements and level of satisfaction with a specific application describing the level of user's perception of an application. QoE consists of a number of influencing factors that must be considered altogether. These factors vary from characteristics of the user and users' expectations to technical terms defined with QoS. Each of these factors has their own weight in the final calculation of QoE. However, in order to be able to trade-off power consumption and QoE, a single measurement parameter is required. Since the main aim of this pa- per is to examine the potential trade-offs between energy consumption and perceived quality we consider QoE as a single parameter that can be calculated using available voice and video quality assessment methods. Below these metrics are introduced for voice and video applications.

In this paper we explore interrelationship between end-to-end perceived quality as measured by objective performance metrics and energy consumption spent for encoding and transmission. Both of these two metrics are modulated by two components that are independent of each other, type of the codec and loss behavior of a channel. Below we firstly consider these independent factors and then describe possible trade-offs and dependencies between perceived quality and energy consumption.

In the rest of this chapter description of the factors involved in the trade-off is provided. Section 2.1 discuses Media compression algorithms examined in this thesis. In section 2.3 QoE and two factors which lead to its degradation will be introduced. It is necessary to mention that term QoE here is limited to Peak Signal to Noise Ratio (PSNR) and Structural SIMilarity (SSIM) as objective measurement methods in video codecs and Mean Opinion Score (MOS) factor as subjective measurement in audio codecs. Finally, discussions regarding energy requirements for transmission and coding phases are provided in section 2.3.

2.1 Media Compression

Compression algorithms are used to reduce bandwidth requirements of media applications. However, at the same time compression brings a number of additional problems to network designers. First of all, compression makes media vulnerable to network impairments such as packet losses and delays. Secondly, modern compression algorithms, especially those, used for video information, are relatively complex involving multiple stages with complicated computational algorithms. It is usually the case that the higher the compression ratio of the algorithm the more complex the algorithm is. Further, compression affects the perceived quality both directly and indirectly. Considering video information as an example, the compression process removes spatial, temporal and psychovisual redundancies encoding the most important parts. It implies that the amount of actual information per data unit in a new compressed bitstream increases.

2.2 Quality of User Experience

Providing good perceived quality is crucial for commercial success of any service provider. Very often this question is reduced to using the most complex codec providing the best possible compression and resulting in the best possible "after-encoding" quality. This approach may indeed result in better perceived quality, bandwidth savings, and less energy required for transmission. The latter two factors are due to injecting less traffic into the network. However, is not as straightforward as its seems at the first sight as more complex codecs imply more energy for encoding, higher encoding delays, and traffic stream more sensitive to packet losses. The latter two factors may completely neglect those gains we get using a complex codec while the first one may neglect energy gains we expect due to better compression. Thus, providing the best possible QoE at any given instant of time is a complex task that depend on many other network and application characteristics. In this project we will address these trade-offs identifying the best possible choices for a given environmental conditions and provide the prototype of the system for adaptive on-line optimization of energy consumption and QoE for media applications.

There are also so-called side-factors affecting the choice of a codec for media applications. For example, high energy consumption of a service may affect its usage by customers in wireless networks decreasing the revenue of service providers and network operators while high throughput requirements may not always be supported by the network. We do not consider these factors.

2.2.1 Speech compression

Quality of VoIP service is usually evaluated at the application layer using specific tests developed for assessing the perceived speech quality. To perform these tests a number of methods have been suggested in the past. We distinguish between subjective and objective tests. Those tests that involve surveying humans are called subjective tests. Objective tests are based on deriving application layer performance metrics based on codec- and/or network-related performance parameters. These tests try to provide the required relationship between network performance and subjective QoE metric.

The de-facto subjective metric for evaluating quality of voice communications is mean opinion score (MOS). MOS provides numerical indication of the quality of received media after compression and/or transmission. The value of MOS is a number ranging from 1 to 5 with 5 corresponding the the best possible speech quality. MOS is estimated by averaging the results of a set of subjective tests where a number of humans grade the heard audio quality of test sentences. A listener is required to give each sentence a rating using grades from 1 to 5. The MOS is the mean of all scores set by individuals.

To conserve the amount of required bandwidth, voice information is always compressed before transmission. The compression process itself may incur significant quality degradation setting the higher bound on achievable speech quality. In addition to specifics of voice compression scheme network may also induce significant performance degradation resulting in lower value of MOS. To take this factor into account the so-called E-model has been standardized by ITU-T. This is objective performance model whose results were initially shown to correlate well with MOS grades. According to E-model the psychoacoustic speech quality is defined as a non-linear additive function of different impairments including network-related parameters. The measure of quality is called R-factor and given by

$$R = R_o - I_s - I_d - I_e + A \tag{2.1}$$

where R_o represents noise and loudness in terms of the signal to noise (S/N) ratio at 0dBr point, I_s represents impairments occurring simultaneously with speech, I_d represents impairments that are delayed with respect to speech, I_e is the effect of special equipment, A is the advantage factor. In other words, I_d is the delay of voice signals being generated by a sender to its playback by a receiver. Coding and compression procedure is included in I_s, while network loss-related quality impairments are in I_e. Advantage factor accounts for special environments, where a user may sacrifice the quality with respect to availability of the service. It is used to represent the convenience of being able to make the phone call. The value of R-factor is between 0 and 100.

In order to estimate the quality of voice communications all the components in 3.14 have to be estimated. Assuming A = 0 and given a certain compression and coding algorithm the R-factor becomes function of Id and I_e only. VoIP routes are usually provisioned such that the end-to-end delay impairment factor is always less than maximum tolerable delay (150-200ms). In this case the quality of speech transmission is dominated by I_e. The effect of I_e has been found using extensive subjective tests. However, relying on the first-order statistics of the packet loss process only may lead to different results in terms of the perceived speech quality. The reason is presence of correlation in the packet loss process. Indeed, state-of-the-art codecs are able to cope with single packet losses reducing their effect using extrapolation of the reconstructed signal. When losses occur in batches extrapolation gives unsatisfactory results.

It is important to note that statistical studies did confirm that the packet loss process in the Internet are often not memoryless. There are two underlying reasons for this behavior. It may happen due to time-varying behavior of Internet routes. Yajnik et al. demonstrated this effect and further showed than it can be the cause of packet loss clipping. One possible explanation for time-varying behavior of the packet loss process is limited lifetimes of Internet routes. Approximately 10 percent of commercial Internet routes had lifetimes of few hours or even less resulting in abrupt changes in the packet loss process. Another reason for bursty nature of the packet loss process is the droptail queuing discipline which is still popular in the Internet. It has been shown analytically and using network measurements that under droptail assumption the buffer occupancy process evolves in cycles resulting in packet losses occurring in bursts.

The underlying reason for this behavior is presence of feedback controlled TCP connections. Bursty behavior of packet losses has also been reported for wireless access networks. Although usage of OFDM, interleaving mechanism, and AMC try to ensure that that memory of the channel process does not propagate to higher layers, we may still observe batched losses during exceptionally band channel quality conditions.

To take into account the effect of packet loss correlation Clark [33] proposed to define two states in the packet loss statistics, namely, loss state and and loss-free state. The proposed model remains in loss state as long as there are no more than m successfully received packets between two loss events. If more than m packets are successfully received the model jumps to loss-free state. The threshold m depends on the type of the codec and its extrapolation capabilities. Whenever state transition occurs, loss-related impairment Ie is computed. The overall loss-related impairment is then computed as average of loss and loss-free states impairments.

Indeed, humans do not instantly perceive changes in quality levels. Changes in state transition are also differently perceived by users. Specifically, transition from loss-free to loss state is usually noticed faster compared to transition from loss state to loss-free state. This effect can be fairly well modeled using exponentially decaying functions with appropriate time constants. Integrating over all possible durations of loss-free and loss states we get the time-averaged loss-related impairments .

2.2.2 Video Compression

Various compression algorithms for video information have been introduced in the literature among which some are used in real-time applications. They can be either lossy or lossless compression with different number of references. Like any other type of media, compression affect quality of video information in two ways resulting in degradation of quality. These techniques will be introduced and discussed in chapter four in details. Here, the used method for calculating QoE after compression and transmission is discussed.

In order to be able to assess the quality of video perceived by human it is required to run a subjective quality evaluation by asking user's perception of the video they have watched. This method is defined as subjective Video Quality Assessment (VQA). However, in most cases it is not practical as it is highly time consuming and expensive. More over, because of involvement of human and different expectations and ideas of users about video streams it might suffer from lack of accuracy. However, these methods make a valuable database to generate a precise objective evaluation algorithm.

Performing above-mentioned subjective tests consists of a set of steps which must take place. One important task is to choose an appropriate testing system and environment and keeping that condition in a constant level during the whole process of experiment. The reason why this is important relates back to the inherent of video quality assessment in which different parameters are important. For example lightening system of the environment will change the opinion of a user while all other features are kept identical. Source of video must also be chosen and do not change during the whole process of evaluation. This can be done by choosing a number of video sequences and performing tests on them. The order of playing video sequences is also important during assessment and must remain unchanged. Because of wide range of scores given by different users, number of evaluators must not be small or the amusement won't be appropriate.

For objective measurement purpose, two metrics have been considered in this study and each metric has been calculated for four different YUV sequences to ensure integrity of result. These two metrics and their calculation methods are introduced below.

PSNR

PSNR is mostly represented in logarithmic decibel scale and is used to calculate the ratio between the power of the original sequence and the erroneous sequence after lossy compression process. Higher PSNR value means that coded sequence has a better quality. In order to measure PSNR, one must calculate Mean Square Error (MSE) between original and compressed video sequence first. MSE is defined as:

$$MSE(I,J) = \frac{1}{mn} \sum_{i=0}^{m-1} \sum_{j=0}^{n-1} [I(i,j) - J(i,j)]^2 \tag{2.2}$$

Then, having MSE for a number of frames of a sequence, PSNR will be:

$$PSNR(I,J) = 10 * \log_{10}(\frac{255^2}{MSE}) \tag{2.3}$$

$$PSNR = \mathrm{M}ean(PSNR(I,J)) \tag{2.4}$$

SSIM

SSIM is well-known measuring method used to calculate similarity between two images. In objective quality measurement methods, SSIM is the full reference metric which is a improvement to traditional methods like PSNR and mostly considered as a metric which closely represents Human Visual System (HVS) quality assessment [13].

To calculate this metric, according to what has been done in [14] we consider two non-negative images aligned together and name them x, and y. let consider μ_x and μ_y as the mean of x and y respectively, σ_x as the variance of x, and σ_{xy}^2 as the covariance of x and y. Then three components luminance, contrast, and structure comparison would be given as follows respectively:

$$l(x,y) = \frac{2\mu_x\mu_y + C_1}{\mu_x^2 + \mu_y^2 + C_1} \tag{2.5}$$

$$c(x,y) = \frac{2\sigma_x\sigma_y + C_2}{\sigma_x^2 + \sigma_y^2 + C_2} \tag{2.6}$$

$$s(x,y) = \frac{\sigma_{xy} + C_3}{\sigma_x\sigma_y + C_3} \tag{2.7}$$

And finally, SSIM would be:

$$SSIM(x,y) = l(x,y)c(x,y)s(x,y) \tag{2.8}$$

It must be noted that this formula corresponds to single-scale structural similarity

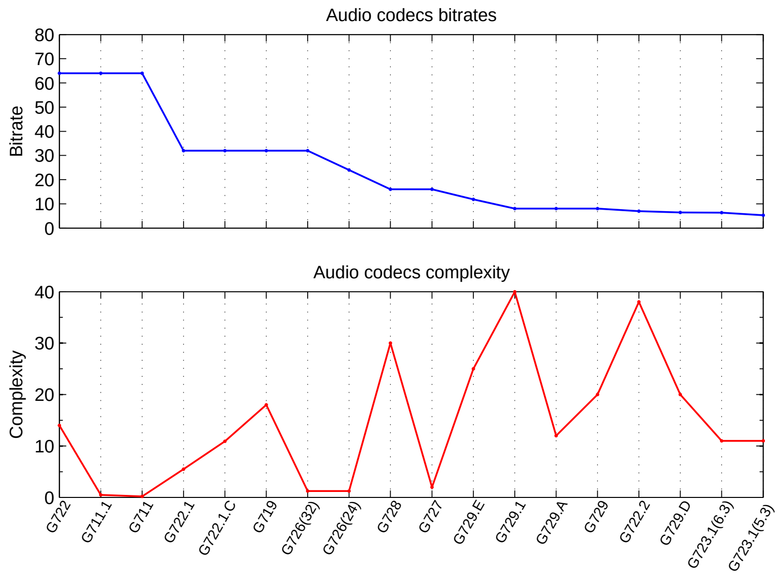


Figure 2.1: Complexity and Bitrate for different audio codecs

and in [15] method for calculating multi-scale SSIM is studied. In this project, PSNR and SSIM for four different sequences foreman, news, Akiyo, and mother and daughter based on 4 different Quantization parameters 20, 28, 30, and 33 are calculated. For this reason, JM v.15.0. which is the H.264 reference codec has been used. More details would be presented in chapter 4.

2.2.3 Encoding QoE Degradation

The encoding procedure is the first factor affecting perceived quality of media applications. The higher the compression ratio (raw rate to compressed rate) the worse the perceived quality, e.g. picture or heard voice quality. This trend is known to have a complex behavior but in most cases can be sufficiently well modeled by exponential function. A very important effect of codecs is that the more the compression ratio is the more information is carried in data units. In network environment it means that the loss of a packet generated by a codec with higher compression ratio will have more negative effect on the perceived quality than that from the codec with lower compression ratio. Finally, the delay incurred by the encoding procedure is inversely proportional to the compression ratio, i.e. the higher the ratio is the more delay is incurred. This effect is also expected to have exponential behavior.

This issue is studied for both video and audio codecs. In Figure 2.1 the ratio can be shown by the final output rate of codecs. Data rate here is kbps and complexity of codecs are measured in Million Instructions Per Second (MIPS). Higher bit rate demands higher bandwidth but less complexity. Encoding is a lossy compression procedure and keeping more part of original data means less compression and higher

QoE. As it can be seen in this figure, the normal trend is that codecs with higher complexity impose less rate on the network. However, at some points this normal trend changes and a codec with higher complexity has slightly higher data rate than others. This feature is because of more complex error concealment algorithms used in the codec which tries to improve QoE. This improvement, requires redundant data to be introduced.

2.2.4 Network QoE Degradation

The second factor contributing to the end-to-end perceived quality degradation is the network. There are two major network characteristics affecting the quality of real-time media communication. These are packet loss and delay processes experienced by individual media sources. For any packet, there is a strict deadline for its arrival to the receiver and, if a certain packet is late, it is considered to be lost. As a result, meeting the delay requirements in the network is essential for adequate performance of real-time media applications.

Nowadays, delay requirement are almost always satisfied. The reason is that delay performance of the Internet and wireless access networks significantly improved over the last decade. In fact, there is some evidence that in late 90s beginning of 2000s end-to-end delay performance of the Internet improved two times over two four years time spans, i.e. 1998-2002, 2002-2006.Now, we are perfectly below the maximum allowed delay of 150-200ms for VoIP and videoconferencing applications. For real-time video applications such as video streaming delay requirement are not that strict due to pre-fetching delay often introduced at the receiving side. This delay usually introduced by implementing the play-out buffer and delaying the beginning of a video by several seconds allows some packets way later than 200ms and still be usable for playing. Due to these reasons in what follows we concentrate on the loss process only.

Nowadays, the loss process is considered to be the dominating source of performance degradation of media services. It is important to note that even if pre-fetching delay is introduced for streaming service lost packet are not retransmitted due to usage of RTP/UDP protocols at the transport layer. Thus, even when delay requirement of an application is satisfied, losses may restrain acceptable operation of the service. This is especially important for wireless channels, where even in presence of AMC and ARQ mechanisms some erroneously received bits propagate through the protocol stack causing loss of protocol data units at IP and higher layers.

It is known from empirical studies that the perceived quality may degrade as fast as exponentially when packet loss rate increases linearly for both video and voice. However, the exponential constant may vary widely and usually depends on the compression ratio, i.e. the more information is carried out in data unit the worse the quality degradation is in presence of losses. The second-order properties of the packet loss process (memory) also have profound negative impact on perceived quality. Taking H.264 video codec as an example, we will see in what follows that, for a certain packet loss rate correlation could make the difference between very bad and very good perceived quality.

2.3 Energy Requirements

The amount of energy spent by a sending side of an application for providing service can be divided into the following three components: (i) energy required to maintain the application state, (ii) energy required for encoding and (iii) energy required for transmission over the air interface. The first component can be considered as constants for a given software/hardware configuration irrespective of the type of the codec used for transmission. Other components may however vary.

The reason why energy is important in this context relates to the rapid growth in portable devices. They are capable of connecting to the Internet through cellular networks or other wireless access methods like WiFi. This trend towards multimedia applications using portable devices has been the main focus of many studies in recent years.

Bin et al [16] investigated different requirements for multimedia applications and then categorized them and discussed limitations. These limitations can be the screen size of the hand-held devices such as smart phones that is partly studied by [17] and [18]. Processing power limitations, communication issues, and energy requirements are among other source of limitations. In the following subsections, energy related issues are explained in more details.

2.3.1 Energy for encoding

The energy spent for encoding varies with the type of the codec and its special features. Considering H.264 video codec as an example, the complexity of encoding and, as a result, the energy required for encoding depends on the targeted resolution and perceived quality via a set of internal parameters, e.g. quantization parameter. The higher the resolution and targeted quality the more energy is required for encoding. At the same time the throughput required by the bitstream increases. The latter dependency is expected to be non-linear implying that with the increase of the compression ratio perceived quality may degrade slower compared to the increase in throughput requirements. Investigations done in this project show that the range of encoding energy for these codecs varies from below 1 mW per packet to even 12 mW in some cases. For more details see Chapter 3.

Figure 2.2 shows encoding power of G729 Annex E audio codec released in 1998.

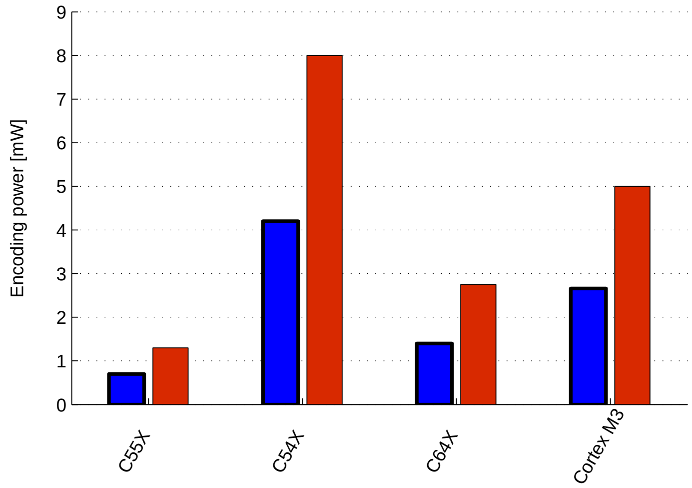


Figure 2.2: Encoding power in mW of G.729.E (red bar) and G722 (blue bar) audio codecs using four different processors.

This codec uses Conjugate Structure Algebraic Code-Excited Linear Prediction (CS-ACELP) coding algorithm and is presented in red. G722 codec released in 1988 which is a sub-band codec is represented in blue in the same figure. This one divides 16kHz bands into 2 sub-bands, each coded using Adaptive Differential Pulse Code Modulation (ADPCM). It can bee seen that energy consumption for all processors in G.729.E is alost double G.722.

In case of video sequences, considering H.264 as an example, the complexity of encoding and, as a result, the energy required for it depends on the targeted resolution and QoE. In this thesis QoE of video sequences will be defined in terms of PSNR and SSIM. The mentioned complexity can be manipulated via a set of internal parameters, e.g. quantization matrix, EC algorithm used, The profile which is chosen for encoding, etc.

2.3.2 Energy for transmission

The energy required to transmit a media bitstream over a wireless channel depends only on the rate of the codec and type of access technology. However, the rate of the bitstream is a function of the targeted perceived quality set for encoding. In principle, the least amount of energy is spent for transmission when the highest possible compression is used and vice versa.

However, as we already highlighted above this dependency is non-linear, i.e. when the compression ratio is already significant one needs to spend a plenty of energy to increase compression ratio a bit more. Moreover, different rates often result in different amount of losses affecting the bitstream, i.e. the codec with higher targeted perceived quality would result in higher throughput requirements and lower end-to-end perceived quality due to excessive losses experienced at the air interface than the codec with smaller targeted QoE and, thus, lower throughput and energy requirements. For different states of transmission energy we have:

$$P_{Tx} > P_{Rx} > P_{idle} > P_{sleep} > P_{off} \tag{2.9}$$

That is why one can assume the energy consumption of transmission is mostly during active state of the device. However, the rate of the bit-stream is a function of the targeted QoE. In principle, the least amount of energy is achieved when the highest possible compression ratio is used and vice versa. Transmission energy can easily be formulated according to [22] and [23] in terms of R and R_{Max} as we can see below:

$$P_{transmission} = P_{active}.\frac{R}{R_{Max}} + P_{inactive}.(1 - \frac{R}{R_{Max}}), \tag{2.10}$$

where R is data rate and R_{Max} is the throughput of the device.

Lower R means lower transmission ratio and obtaining lower R is dependent to the codec and its configuration. For example, in JM configuration file one might find some parameters affecting R using them find the most suitable rate for the application considering other side-effects of that parameter. One might reduce rate but simultaneously double complexity and as a result energy consumption for encoding. In this scenario, instead of any decrement in energy consumption, the final result might show increment in consumption by decreasing rate.

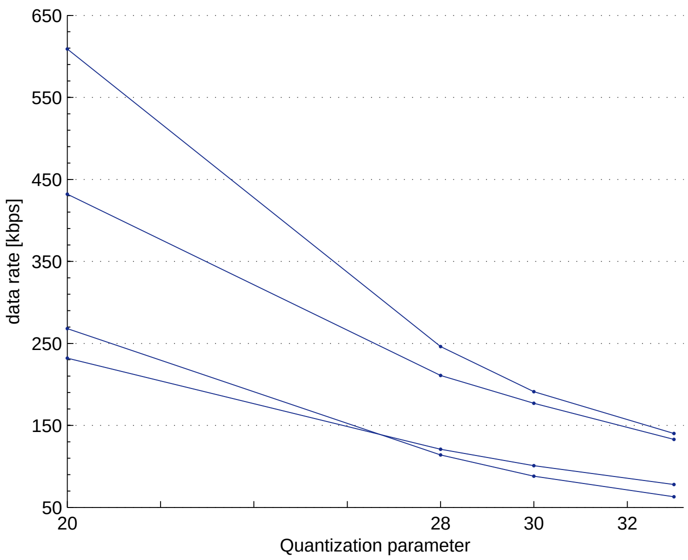


Figure 2.3: Data rate versus Quantization parameter

Another factor which can be used is quantization parameter discussed earlier. Increasing QP value surely decrease output data rate and this decrement in the rate can be considerable as it is depicted in Figure 2.3. There, the effect of QP on data rate is represented for four different sequences all in Quarter Common Intermediate Format (QCIF) whose resolution is 176*144 pixels. The highest curve corresponds to Foreman sequence and the following curves are for News, Mother-and-Daughter, and Akiyo respectively.

As it can be seen there, increasing quantization steps can decrease the rate and as a result energy required for transmission. However, as it is discussed before, higher QP values result in lower PSNR and SSIM of the output sequence. Having in mind the requirements of the application, one can find the best case accordingly.

2.3.3 Energy for different technologies

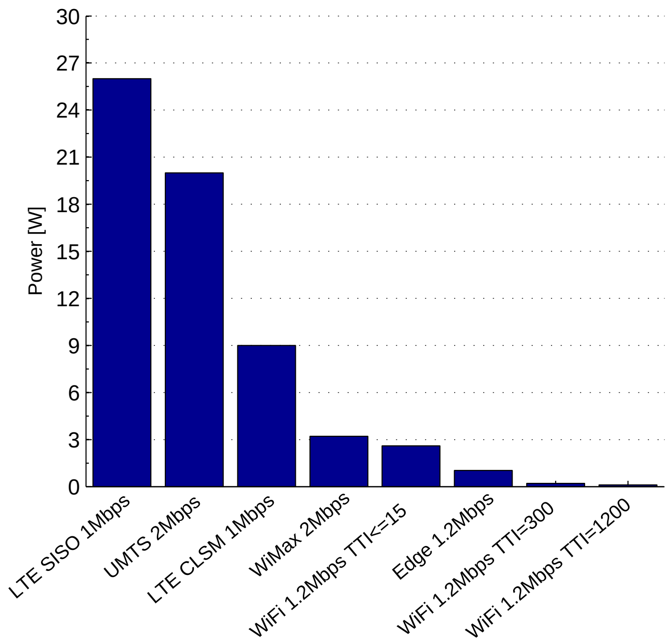


Figure 2.4: Power consumption of different transmission technologies.

Another factor which plays a significant role in energy consumption related studies is the wireless technology chosen for transmission. Although in some situations it might not be possible to choose the technology, the considerable difference in the amount of energy required for transmission per packet, makes it crucial to take this factor into account as well.

The range of transmission energy differs from 26 Watt per packet in 1 Megabits per Second (Mbps) Single Input Single Output (SISO) Long-Term Evolution (LTE) to 0.1 Watt per packet for 1.2 Mbps WiFi while Transmission Time Interval (TTI) is set to 1200s. This huge difference can make a network the main source of energy consumption or an energy efficient one, while producing the same QoE in the same rate. Nowadays, when handheld devices are often equipped with various air interfaces (e.g. WLAN and LTE) choosing the most appropriate technology according to the data rate required and the range of wireless network is one possibility for optimized performance of media services on the move.

Figure 2.4 proves that choosing an appropriate transmission technology, if possible, would have a non-negligible impact on the total energy consumption. The highest energy corresponds to LTE SISO whose rate is 1Mbps and then Universal Mobile Telecommunications System (UMTS) cellular system whose power consumption is 20 Watt at the rate 2Mbps.

3. AUDIO CODECS

The main focus of this chapter is on trade-offs between previously introduced components in audio codecs . The reason why it is important to compare these factors in these codecs and the applications which use them are examined in detail. Since energy required for encoding in case of audio seems negligible compared to energy for transmission, one might say that it is not tenable trading-off them. However, as it will be shown, there are some codecs whose encoding energy is more than energy required for transmission.

This chapter is structured as follow. Section 3.1 introduces VoIP applications and challenges in this area. Different algorithms are used for compression in different codecs. each of them have their own advantageous and disadvantageous. Section 3.2 introduces these methods and compare them. After two primary sections of this chapter, section 3.3 compares all these factors and performs desired trade-off in audio codecs.

3.1 VoIP challenges and protocols

This section, tries to study three important factors of VoIP systems with a focus on voice codecs and their features. As it is already discussed, compression, energy consumption and QoE are of high importance. In order to be able to use any VoIP application, one must decide which codec to choose and choosing this codec require precise considerations. Optimizing power usage, bandwidth consumption, and QoE are among these considerations. However, at first it is necessary to see what is VoIP, other alternatives and why to use or do not use VoIP services.

Traditional communication system enabling exchanging voice data i.e. Public Switched Telephone Network (PSTN) and the well-known VoIP are two architectures which play an important role in daily communications. In order to distinguish between advantages and disadvantages of either of these systems, reliability, simplicity of the connectivity, stability, and of course their applications must be taken into account.

Current condition of the Internet and issues regarding mobility, bandwidth available, congestions and losses, and other non-ideal limiting factors make PSTN communications more stable and reliable. On one hand, there is VoIP and its wide range of applications. On the other hand, expensive and time consuming phase of installing circuit-switched networks and their cabling complexities. all thes, made traditional PSTN companies aware of the need to change [25] or they might suffer from a huge a financial problem.

Various studies is already done to compensate network defects that affect quality of VoIP sessions negatively. Zvezdan et al [26] r investigated PSTN networks and figured out the optimized number of circuits necessary for communications. Using them, they presented a software which is capable of estimating required bandwidth. Yuan [27] introduced an alternative solution to replace H.323 standard to conceal its weak points. There, he presented Session Initiation Protocol (SIP) as a signaling system using a comprehensive study from the features of this protocol to its architecture and inter-working of this VoIP network with traditional PSTN.

3.1.1 QoS weak-points

The original application of IP networks was to provide an environment for non-real-time data services. The reason is that these networks operate based on best-effort algorithm. That is why guaranteeing specific quality of service for real-time applications demand additional protocols and considerations. According to this inherent characteristic of IP networks, VoIP providers must introduce and take advantage of new mechanisms. Influencing factors on degradation of quality are listed below:

Delay

Delay is one important factor which degrades the quality of VoIP applications and is introduced by both the codec used in the application and impairments of the channel. According to these different resources of delay, one can divide it into network delay, jitter delay, and codecs' processing delay. The latter, is studied in this project in more details and the result of these studies, is gathered into table 3.1. This table, is a good reference to have the delay of a wide range of ITU-T voice codecs all in one place.

Table 3.1: Processing delay of voice codecs [ms].

G.711.D 12.81	G.722 4	G.722.B 12.31	G.711.1 11.87	G711 0.125	G.718.B 49.62	G.729.1.E 55.69
G.722.1 40	G.722.1.C 40	G.719 40	G.726 0.125	G.723 30	G.728 0.625	G.729.E 15
G.729.1 48.94	G.729.A 15	G.718 42.87	G.729 15	G722.2 25	G.729.D 13	G.723.1 37.5

Packet loss

One problem that considerably affects the quality of service of VoIP applications is packet loss. Although there are some procedures to conceal the effect of packet loss on the quality of the voice, packet loss rates over 10 percent, will affect the quality noticeably. There might be some conditions that they might result in the dropping of the voice session.

Although packet losses may occur due to many reasons, mechanisms used in the Internet these days, makes the probability of these losses happening as a result of noise or other impairments very low. The main source of degradation in QoS due to packet loss is those created by congestion. In order to be able to study the impact of this kind of loss, one must consider the correlation between packet loss and delay on the Internet. The reason of this phenomena is that one sign of possible congestion in the network is increment in the delay (one-way delay) as this shows that there is at least one bottleneck in the network whose queue is getting overloaded.

Some applications have an intelligent system which adapts the rate of the application based on the packet loss rate of the network. Although, this feature might help improve the quality of service considerably, the application must be able to distinguish between different kinds of packet losses. If it is just a random packet loss, decreasing the output rate might not be a good idea. However, packet losses due to congestion are mostly continuous to some extent. In that condition, decreasing the rate is a good way to improve the quality, but in some cases, it might be a crucial approach to keep the session alive.

Bandwidth

Available bandwidth during a VoIP session may vary due to many reasons. Because The consequence of these changes in the bandwidth is that reliability of the quality of VoIP conversations is not guaranteed and change in time. Another effect of bandwidth problems correspond to the protocol used in VoIP applications. UDP does not guarantee delivery of packets and any drop in packet which happen as a result of the available bandwidth affect quality of the voice negatively.

One approach to deal with this problem is to some how reserve bandwidth for VoIP traffic. One of these approaches which has newly been introduced is presented by Luca [29] in which he proposed a bandwidth reservation scheme for VoIP traffic by adopting neural technique. To achieve this reservation, the only item that should be monitored and calculated is packets size and then using some mathematical formulas and implementing the model on some cost-effective devices like FPGAs it is possible to reserve bandwidth. This way, one can improve QoS of VoIP application drastically.

Security

Security issues are always of high importance for any application in the Internet. In VoIP applications, intruders may be able to listen to other's conversationsl. On the other hand, they might get access to their voice mail or even find out their ingoing and outgoing calling numbers [30]. For example, one might place a packet grabber on the network while has gained access to the transmission protocol over IP-network and perform eavesdropping as it is done in PSTN systems.

There are a number of security concerns for VoIP applications each of them targeting one or more aspects of the system. Denial of Service (DoS) attacks target Availability of VoIP applications. Getting access to the application and its outgoing and incoming calls might make intruders capable of threatening the confidentiality, integrity and availability of the VoIP system all the same time by making the connection to a third party for example.

In the literature, a number of ways have been investigated to face security issue in VoIP applications. David et al [31] after introducing different attacks in detail and the protocols on which these attacks may impact, presented some solutions for defending security issues. They have shown that one way to make VoIP systems safer is to separate VoIP and data traffic. The other way might be authenticating configuration and/or the signaling protocol and finally, encrypting media might also help improve security of VoIP systems.

Other factors

In addition to above-mentioned factors which affect QoS of VoIP systems negatively, there are a number of other issues which will have a bad impact on these systems. For example, In poor network conditions, the packets might arrive out of order and this impairment in the delivery of packets might influence the quality of the system. However, this issue might be solved easily as voice packets mostly have a time stamp using that and by using a short buffer list it is easy to reorder packets again.

QoS degradation due to lossy compression algorithms is one thing which must be considered. All voice codecs transform analog signals to digital ones and also decrease their size in order to be able to be transmitted over network. These compression algorithms lower the quality of the output which can be presented by constant I_e in the E model which will be presented for different codecs in the following sections of this chapter.

3.1.2 Standards and protocols

One of the main organizations which standardizes protocols for VoIP applications are Internet Engineering Task Force (IETF) that is an organized activity of the

Internet Society. This community declares how Internet protocols and how the Internet works. The other organization is the ITU which manages global telecom networks and services.

Each VoIP system consists of a number of components co-operating with each other. These components include various gateways to control media and signaling procedures and also a call agent. Synchronization of all these components and handling their issues require different set of rules and protocols.

There are different protocols co-operating to make VoIP services possible. Megaco, SCCP, RVP, MIMO, MGCP are some of these algorithms that are beyond the scope of this project and introducing them might be confusing. However, there are two main protocols i.e. SIP and Session Description Protocol (SDP) which are introduced in this section.

SIP

SIP is a signaling protocol operating at application layer and is client-server based protocol. Using SIP makes end user systems capable of providing a set of services. It might seem so simple for users to start a session with other users by pressing only one button. However, the first problem for session initiation which is solved by this protocol is to map the user's account id which in most cases is user's Email address or phone number with its machine.

SIP does this by sending invitation messages from first user's machine to its provider server and then another message to the second user's server and finally the last invite request to the second user's machine. These messages can be sent either over UDP or Transmission Control Protocol (TCP) and at least must contain the address of the second user and the protocol that makes target machine aware of the parameters of media session which sender requests for the VoIP session. The latter will be introduced in more detail in the following subsection.

Using SIP it is possible to send starting conference sessions, user's multiple devices accessibility for one single user for example from laptop and smart phone, authenticating caller and identifying calling number. Additionally, it enables Internet telephony gateways which are connected to PSTN parties call each other. Another important feature of SIP is its capability to work with other signaling protocols which provide a new set of services.

In addition INVITE messages, there are also a number of other types of messages which makes all these services possible. ACK message is transmitted to inform the other party of the confirmation of last response. To terminate a call it is required to send BYE. If the user wants to cancel ringing before starting conversation, SIP sends CANCEL message. There might be some cases in which having a specific codec or service is not supported by the other party and it is shown in OPTIONS message used by this protocol. It must be mentioned that all these are in the request messages group. Request messages' structure is shown in table 3.2

Table 3.2: SIP REQUEST message structure.

Method	Request URL	SIP version

SDP

Communication end points need to be at their most effective condition during a communication session. SDP does this by introducing a set of rules in the set up phase of sessions. For this purpose, SDP describes multimedia sessions containing information about session time, data and name, address of end points, and the format of the data. SDP messages are transmitted over UDP and their payload is text.

3.2 Codecs compression technologies and their applications

Three factors introduced in Chapter two as the trade-offs would be studied in this subsection for different audio codecs, processors, and transmission technologies. For this reason, ITU-T codecs have been chosen and studied.

Different codecs are intended for different applications based on the coding algorithm used in them and also other factors such as their quality, bandwidth usage, payload size and so on. These main applications of codecs have been gathered and will be introduced in this study. However, they might be used in other applications in some case but those which are provided here are their main usage.

In order to have a better classification, audio codecs are divided into two general groups based on their Packet Per Second (PPS) feature. PPS as it is clear from its name is the rate of generation of audio packets per second and can be calculate as equation 3.1. Using PPS, one can calculate packet duration defined as the time it takes between generation of first bit of packet i-1 and the first bit of packet i.

$$P_s = \frac{R}{L_s} \tag{3.1}$$

$$P_d = \frac{1}{P_s} \tag{3.2}$$

Where P_s is PPS, R is the bit rate, L_s is payload size, and P_d is packet duration. Table 3.3 contains codecs whose PPS is more than 50 and table 3.4 corresponds to codecs with PPS over 50. As a general trend codecs which are introduced after the year 2000 have PPS values over 50 and for older codecs, this value is 50 or below. However, G729.A is an exception in this regard.

To be able to summarize technologies in the most optimized way, they are classified into different numbers and the corresponding number of each codec is represented in the table. For this reason, first introduce these numbers as below:

1. ISDN, video conferencing, Voice over Packet Network (VoPN)
2. 3G Wireless
3. VoIP
4. VoATM, ToIP, IP Phone, Private Networks
5. Teleconferencing, Telepresence system
6. WiFi phones VoWLAN, Wireless GPRS EDGE systems, Personal Communications, Wideband IP telephony, and Audio and Video Conferencing
7. multimedia
8. Speech recording and archiving, Digital circuit multiplexing equipment
9. Audio streaming
10. G.729E is a higher rate version of G.729 and is designed to provide higher quality for background noise conditions, music, and tandems. This mode is well suited for music, and it has greater complexity than the original G.729 codecs
11. G.729 Annex A offers the best complexity/quality ratio in the industry
12. Annex D supports multiple bit rates, 6.4/8kbps. Applications include Voice-Over-IP, Voice-Over-ATM, VAD, video conferencing systems, multimedia, store/forward and satellite/wireless communications.

MDCT transform description

As it can be seen, in the description part of the table there are some new items which must be defined first. Modified Discrete Cosine Transform (MDCT) is a variant of Fourier transform whose difference with other transforms is in the number of outputs after process. MDCT operates on the input and the number of components of the result would be half the input. In addition it must be noted that input must be real number. If we consider N real numbers as the input, output after transformation will be N/2 real numbers. The formula of MDCT is:

$$X_F = \sum_{n=0}^{N-1} x_n \cos\left[\frac{\pi}{N}(n + \frac{N+1}{2})(F + \frac{1}{2})\right] \tag{3.3}$$

Table 3.3: Audio codecs applications and their description with PPS over 50.

Codecs	PPS	Applications	Description
G.711.1	100	3,6	MDCT, A-law, μ-law
G.711.1.D	240	5	G711.1, BWE, AVQ
G.722.B	800	5	G722, MDCT, BWE, AVQ
G.718.B	56	5	G.718, Sinusoidal coding, MDCT
G.729.1.E	56	5	G.729.1, Sinusoidal coding, MDCT
G.722.1	100	1	low frame loss
G.722.1.C	100	5	MLT
G.719	100	5	MDCT,FLVQ
G.729.A	100	11	CS-ACELP
G.722.2	80	1,2	multi-rate wideband ACELP
G.729.D	80	12	CS-ACELP

A-law and μ-law

Analog signals first need to be digitized to be applicable in digital communication systems. For this purpose and to improve the efficiency of linear encoding , a set of methods are available from which we consider A-law and μ -law here. In fact these algorithms are used to optimize the digitization process. A-law does this by

$$F(x) = \operatorname{sgn}(x) \begin{cases} K|x| & |x| < \frac{1}{K} \\ \frac{1+\ln(K|x|)}{1+\ln(K)} & \frac{1}{K} \leq |x| \leq 1 \end{cases} \tag{3.4}$$

Where K is the compression parameter which is equal to 87.7 or 87.6. In addition to this A-law which is used in European countries, there is a μ-law utilized in North America and Japan. The purpose of latter is the same as A-law. μ law has two types. for the continuous form the formula is:

$$F(x) = \operatorname{sgn}(x) \frac{\ln(1+\mu|x|)}{\ln(1+\mu)}, \qquad -1 \leq x \leq 1 \tag{3.5}$$

BWE

Human ear is not equally sensitive to high and low band distortions in the signal. In fact, ear cannot detect distortions in high band very accurately. Advanced codecs, make use of this property to compress digital bits while keeping the quality of encoded audio in a good condition. G.729.1, G711.1 Annex D, G.722 Annex B and some other codecs take advantage of this procedure.

Codecs which use Bandwidth Extension (BWE) encode the lower frequency band of the signal and then using correlations between lower band and the higher one try to predict the latter. Although this might not be a perfect guess, lack of precise detection of higher frequencies in human ears, make this prediction a reasonable approach.

AVQ and FLVQ

In lossy data compression algorithms, in order to change a multidimensional vector space, which require a high storage space, to a vector with lower dimension demanding less memory to store data, vector quantization techniques are used. For this reason, mostly a codebook is required for this transformation to be done. Algebraic Vector Quantization (AVQ) is a kind of procedure in which the codebook does not need to be stored and as a result the efficiency of this algorithm is even better. Fuzzy Learning Vector Quantization (FLVQ) is another form of vector quantization algorithms in which codebook is obtained through a parameter called gradient-descent learning.

Sinusoidal coding

It is possible to represent an original signal as a summation of various sinusoids. In sinusoidal coding, instead of the signal, parameters of this sinusoid is transmitted and at the receiver side the original signal can be reconstructed. There are a number of schemes for low-rates sinusoidal audio coding both in Time Differential (TD) and Frequency Differential (FD). Different schemes have different delays and frame lengths.

MLT

Modulated Lapped Transform (MLT) is mostly used to make it possible to implement block transform coding in audio compression process in fact it is used to have the analogue audio signal in its frequency domain or time domain representation. In block transformation a DCT would be applied to the data of length M to decorrelate it and then the M coefficients would be encoded.

ACELP

Voice Age corporation introduced a technology called ACELP which nowadays is widely used in different codecs. In ACELP, two different filters are used which are Time Variant (TV). Using these two, speech signals with i samples can be synthesized and scaled. The transform function of two filters in z transform are presented below.

$$\frac{1}{\mathrm{H}_1(z)} = \frac{1}{1 - gz^{-t}} \tag{3.6}$$

$$\frac{1}{\mathrm{H}_2(z)} = \frac{1}{1 + \sum_{n=1}^{N} a_i z^{-i}} \tag{3.7}$$

Where g is the gain parameter which the scaling process would be done according to that. CS-ACELP used in G729 annex A and D is Conjugate Structure-ACELP and is designed to operate with an appropriately band-limited signal [32].

ADPCM

It sometimes happens that the minimum bandwidth required for transmission must be decreased further while the Signal to Noise Ratio (SNR) remains at the same level. For this purpose, codec must be able to increase quantization step to do so. However, this procedure may affect output quality negatively. This can be achieved using ADPCM.

Getting familiar with different compression technologies used in codecs, it is time to see technologies and the description of those codecs in which PPS is lower than or equal to 50. Table 3.4 presents these codecs and their features.

Table 3.4: Audio codecs applications and their description with PPS lower or equal to 50.

Codecs	**PPS**	**Applications**	**Description**
G.722	50	1	ADPCM
G.711	50	6	A-law, μ-law
G.726 r32/r24	50	1,2,7	ADPCM
G.728	33	1,3,7,8	G.718, CELP
G.727	50	6	ADPCM
G.729.E	49	10	CS-ACELP
G.729.1	50	3,4	
G.718	50	3	ACELP, MDC, AVQ
G.729	50	1,3,7	CS-ACELP
G.723.1	33	3	

3.3 Trade-offs in audio codecs

Previous sections, performed a deep investigation over different aspects of audio codecs.compression algorithms used in a wide range of frequently used codecs and their applications were examined there. This section, studies trade-offs introduced at the beginning of this study i.e. compression, QoE, and energy efficiency.

3.3.1 Compression

Different codecs encode and produce voice data at different rates. Higher complexity of codecs might result in lower data rate which reduce energy required for transmission. However, running those codecs will increase computational complexities which means higher processing energy. On the other hand, compression algorithms used by codecs have a direct impact on the QoE of the output as well.

Comparing codecs compression ratios makes it necessary to consider other factors than data rate as an indicator of compression efficiency as well. Payload is one of those factors. Payload of the codecs is introduced both in milliseconds and bytes. In addition, calculating bandwidth consumption is another process which would be possible when the payload of the voice data is in hand. Band width calculation would be different as different technologies impose their own header. The general equation which is used here for calculation of bandwidth is presented below:

$$S = H + P \quad [Bytes] \tag{3.8}$$

$$B = S * PPS \quad [KBps] \tag{3.9}$$

Where H is header size and P is the payload of voice packets. In formula 3.9, B denotes bandwidth and S stands for voice packet size. the total voice packet size can be calculated knowing that the packet is transmitted over Ethernet normally or, for example, Compressed Real-Time Transport Protocol (cRTP) is used in transmission. It must be noted that cRTP is not an option while using Ethernet. in normal transmission, the IP/UDP/RTP header would be 40 bytes as follow:

1. IP header=20 bytes (20*8=160 bits)

2. User Datagram Protocol (UDP) header=8 bytes (8*8=64 bits)

3. Real-Time Transport Protocol (RTP) header=12 bytes (12*8=96 bits)

However, there are some cases in which cRTP is used. Using this protocol reduces the 40 bytes of overhead to only 2 bytes. As mentioned before, this technology cannot be used over Ethernet. Another protocols which might be used here are Multi-link Point-to-Point Protocol (MLP) or Frame Relay Forum (FRF). They will add 6 more bytes as layer 2 header. In case of Ethernet, this will be 18 bytes of layer 2 header which also includes Frame Check Sequence (FCS) or Cyclic Redundancy Check (CRC) headers.

To conclude the voice packet size used in the formula 3.8 which is the total packet size, can be calculated using the formula below. According to the type of protocol used, layer 2 header and/or IP/UDP/RTP header sizes might change. As a result the amount of bandwidth usage per call for different codecs in kbps can be calculated as:

$$H = \text{Layer2header}(MPorFRF.12orethernet) + (IP/UDP/RTP) \tag{3.10}$$

$$Layer2header[bytes] = \begin{cases} 6 & MP \\ 2 & FRF.12 \\ 18 & Ethernet \end{cases} \tag{3.11}$$

$$IP/UDP/RTPHeader[bytes] = \begin{cases} 40 & Ethernet \\ 2 & cRTP \end{cases} \tag{3.12}$$

Considering different cases of formulas presented above, bandwidth usage can vary in a wide range. Figure 3.1 depicts the differences in bandwidth usage using different protocols. This study, investigates almost 20 different voice codecs. However, the payload size for these codecs changes between 10 bytes to 80 with steps of 10 and there are also two 160-bytes-payload codecs. This figure takes into account all these values of payloads regardless of the codecs. PPS values are taken from Tables 3.3 and 3.4.

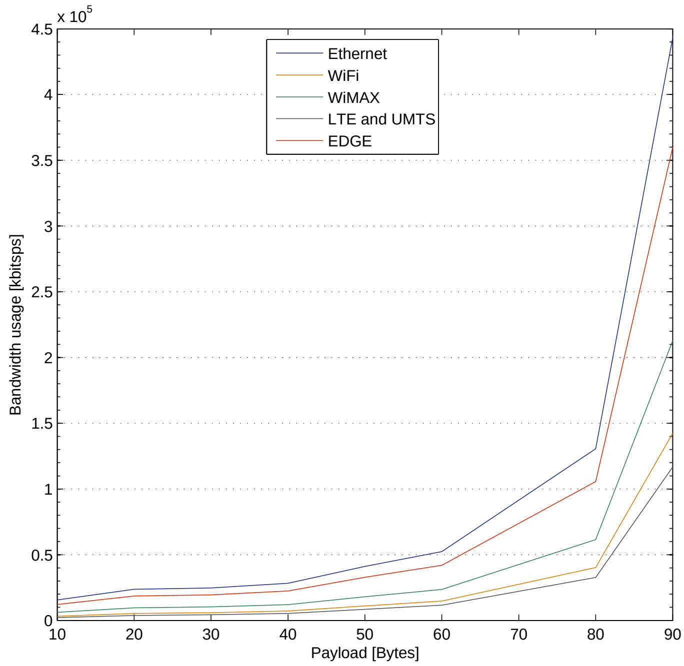


Figure 3.1: Bandwidth usage of audio codecs.

Figure 3.1 shows that using Ethernet, imposes highest bandwidth usage and FRF.12 has the lowest usage as the amount of overhead it adds to voice payload is much lower than Ethernet. All these three curves are some how similar to exponential curve (red one in the figure) and their increment is almost exponentially based on the payload size of the codec.

Table 3.5: Audio codecs Bitrate, payload, Ethernet bandwidth required.

Codecs	**bitrate [kbps]**	**payload [Bytes]**	**payload [ms]**	**Eth. Bw [kbps]**
G.711.1.D	96	50	5	245.76
G.722	64	160	20	87.2
G.722.B	64	10	5	563.2
G.711.1	64	80	5	126.4
G.711	64	160	20	95.2
G.718.B	36	80	20	71.1
G.729.1.E	36	80	20	71.1
G.722.1	32	30	20	86.4
G.722.1.C	32	30	20	86.4
G.719	32	40	20	94.4
G.726	32	80	20	55.2
G.726	24	60	20	47.2
G.728	16	60	20	31.5
G.727	16	40	20	47.2
G.729.E	11.8	30	20	43.2
G.729.1	8	20	20	39.2
G.729.A	8	10	20	39.2
G.718	8	20	20	39.2
G.729	8	20	20	31.2
G.722.2	6.60	10	20	38
G.729.D	6.4	10	20	56.32
G.723.1	5.3	20	30	21.9

Table 3.5 presents all values of different codecs payload sizes both in second and bytes. It also the rates of codecs plus bandwidth required per call over Ethernet. The point is that some codecs have variable bit rates which this also improves their capabilities in lowering bandwidth usage. However, this table only contains one of their data rate and the corresponding bandwidth is based on the bitrate written in the table.

3.3.2 QoE

Recall, that according to E-model R-factor is given by

$$R = R_o - I_s - I_d - I_e + A \tag{3.13}$$

where R_o represents noise and loudness in terms of the signal to noise ratio at 0dBr point, I_s represents impairments occurring simultaneously with speech, I_d represents impairments that are delayed with respect to speech, I_e is the effect of equipment, A is the advantage factor.

Parameterizing the model we see that the advantage factor can be set to 0 as users now get accustomed to wireless voice services. Following Clark the highest possible value of R0 - I_s is set to 94 resulting in reduced expression R = 94-I_d-I_e setting the upper bound on perceived quality. As we discussed in Section 3 the effect of delay can be ignored as most end-to-end paths in the network satisfy delay budget of 150ms. Further, we also highlighted in Section 3 that the loss process of packet can be considered to be memoryless implying that there are no significant grouping of packet losses. It is important to note that all these assumption can be relaxed whenever found inappropriate. However, for this particular study, taking into account special such extreme regimes would result in unnecessary increase of complexity and may hide the main message of the study. Thus, the approximate model now reads as

$$R = 94 - I_e \tag{3.14}$$

The only unknown we have is I_e which is the effective equipment impairment factor telling us how low bit-rate codecs may affect negatively quality. More over, the impact of packet losses whose distribution is random is also included in this component.

The effect of different codecs on the voice quality according to values provided in G.113 recommendation are given in Table 3.6. It must be mentioned that they don't take into account packet losses. In addition to that, MOS value for these codecs are also provided in the same table.

Considering a threshold for MOS value, one can compare different codecs in terms of their QoE easier. Figure 3.2 considers 3 areas of quality evaluation. One is Excellent region which is for MOS values over 4. The other one, acceptable/ Good region corresponds to those codecs whose MOS is between 3 and 4. Finally, unacceptable part which contains only G.727 with data rate 16 kbps whose MOS value is below 3.

3.3.3 Energy Consumption

Total energy consumption is the sum of energy consumed for transmission and processing. Processing power at the sender is the amount of energy a device consumes to run the codec algorithm.

Figure 2.2 proved that energy for encoding using a specific processor may be almost 8 times for the same codec in another DSP. Although choosing between DSPs require considerations more than only energy consumption, they are beyond

Table 3.6: Audio codecs MOS and I_e value.

Codecs	**MOS**	**I_e**
G.711	4.3	1
G.729.E	4.2	4
G.711.1	4.17	0
G.722	4.13	16
G.722.1.c R48	4.10	-
G.722.1 R32	4.09	-
G.722.1 R24	3.98	12
G.729	3.92	10
G.723.1 R6.3	3.9	15
G.726 R32	3.85	7
G.727 R24	3.83	25
G.722.1.c R32	3.80	-
G.729.a	3.70	20
G.723.1 R5.3	3.65	19
G.728	3.61	7
G.729.d	3.60	20
G.722.1.c R24	3.52	-
G.726 R24	3.51	25
G.727 R16	2.84	50

this study and here the only important thing is power consumption.

On the other hand, in Figure 2.4 it is shown that the communication technology plays a non-negligible role in the energy required for transmission. Using 3G LTE has its own benefits. however, in that figure it can be seen that LTE SISO with 1 Mbps bandwidth would consume almost 25 times WiFi 1.2 Mbps whose TTI value is set to 1200.

Considering above-mentioned facts proves that it is not convenient to examine only QoE or compression efficiency of codecs. One of existing correlations correspond to compression ratio. Good effect of higher ratio is in using bandwidth efficiently. However, as it is already mentioned, in poor network conditions, loosing more compressed packets means more loss in information. This is even more important examining handheld devices where lifetime of device depends on the battery. Choosing a wrong codec or an inappropriate technology may result in wastage of battery and make the device working hours without recharging noticeably low.

Encoding power

The energy spent for encoding varies with the type of the codec and its special features. Unfortunately, the actual energy depends on the type of a digital signal processor used for encoding. One of the ways how to provided hardware independent

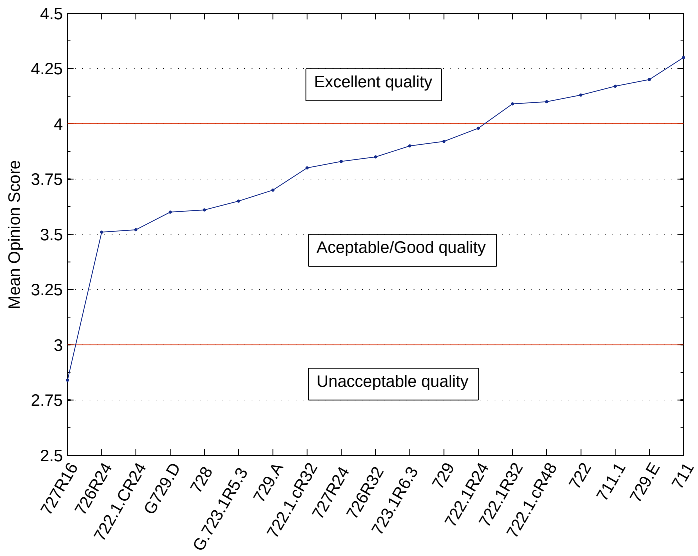


Figure 3.2: Dividing MOS of audio codecs among three different regions of quality.

estimate the complexity it is to calculate the amount of operations required for encoding. Fig. 4 shows raw bitrate and complexity of voice codecs measured in millions operations per second (MIPS), where codecs are sorted in descending order of their bitrates. We see that the general trend is increase of the complexity in response to smaller bitrates. At one extreme there are G.711 and G.711.1 codecs having 64Kbps raw rates and requiring very small processing power. There are exceptionally complex codecs such as G.729.1 and G.722.2 requiring low raw rates. However, there are some exceptions such as G.727, G.729 Annex A, and G.723.1 codecs characterized by rather low rates and moderate encoding power consumptions.

Four different DSPs have been examined in this study. At first it was necessary to have an estimation of codecs processing requirements. Most of voice codecs announce their processing requirements in MIPS, Million Operations per Second (MOPS) and Weighted Millions of Operations Per Second (WMOPS). According to their data sheets the amount of energy they require per MIPS have been extracted. Used processors are listed below:

1. C55x power consumption rate=0.05 mw per MIPS

2. C54x power consumption rate=0.3 mw per MIPS

3. C64x+/66x/674x/64x power consumption rate=0.1 mw per MIPS

4. Cortex M3 power consumption rate=0.19 mw per MIPS

Table 3.7: Energy required for encoding using 4 different DSPs.

Codecs	**MIPS**	**C55x**	**C54x**	**C64x+/66x/674x/64x**	**Cortex M3**
G.722	14	0.7	4.2	1.4	2.66
G.711.1	0.5	0.02	0.15	0.05	0.09
G.711	0.2	0.01	0.06	0.02	0.03
G.722.1	5.5	0.27	1.65	0.55	1.04
G.722.1.C	10.9	0.54	3.27	1.09	2.07
G.719	18	0.9	5.4	1.8	3.42
G.726	1.25	0.62	0.37	0.12	0.23
G.728	30	1.5	9	3	5.7
G.727	2	0.1	0.6	0.2	0.38
G.729.E	25	1.25	7.5	2.5	4.75
G.729.1	40	2	12	4	7.6
G.729.A	12	0.6	3.6	1.2	2.28
G.729	20	1	6	2	3.8
G.722.2	38	1.9	11.4	3.8	7.22
G.723.1	11	0.55	3.3	1.1	2.09

Table 3.7 represents processing complexity of codecs in MIPS along with their corresponding encoding power in different DSPs. As it can be seen, C55x which is from Texas Instruments C5000 series, has the lowest power consumption in comparison with others[34].It is assumed that ARM 7 processors are used in combination with c500 DSP series. For C6000 series, mostly ARM 9 processors will be used. TMS320C6x is an example of c6000 series DSP and TMS320C55x is an example of C5000 family.

Transmission power

It is mentioned that trading power consumption is important and one component of energy consumption resources have been investigated above. however, another important factor in this context is energy required for transmission which would be discussed in this part.

Estimating energy consumed in transmission depends on many factors from the range of coverage to data rate, transmission power shown by T_x, node B, and so on. However, this study takes into account an average but convenient estimation for different technologies.Those technologies which are studied in this paper and a brief definition of them are listed below. These are power consumed in the sending phase.

1. LTE 1 Mbps SISO and LTE 1 Mbps clsm. It must be mentioned that LTE is the newest wireless broadband technology whose access technology is SOFDMA which is multiple access which allows LTE to support variable bandwidths.

Table 3.8: Energy required for transmission over different technologies [mw].

Codecs	**LTESISO**	**LTECLSM**	**WiMAX**	**UMTS**	**Edge**	**WiFi** TTI=1200
G.722	49.5	17.13	3.04	19.04	1.63	0.16
G.711.1	32.86	11.37	2.02	12.64	1.08	0.1
G.711	49.50	17.13	3.04	19.04	1.63	0.16
G.722.1	22.46	7.77	1.38	8.64	0.74	0.07
G.722.1.C	22.46	7.77	1.38	8.64	0.74	0.07
G.719	24.54	8.5	1.51	9.44	0.81	0.79
G.726	32.86	11.37	2.02	12.64	1.08	0.1
G.728	28.7	9.93	1.76	11.04	0.95	0.09
G.727	24.54	8.5	1.51	9.44	0.81	0.79
G.729.E	22.46	7.77	1.38	8.64	0.74	0.07
G.729.1	20.38	7.05	1.25	7.84	0.67	0.06
G.729.A	18.30	6.33	1.12	7.04	0.6	0.06
G.729	20.38	7.05	1.25	7.84	0.67	0.06
G.722.2	18.30	6.33	1.12	7.04	0.6	0.06
G.723.1	21.21b	7.34	1.3	8.16	0.7	0.07

2. WiMAX which is intended for broadband communication and standardized in IEEE 802.16. Fixed WiMAX uses Orthogonal Frequency Division Multiple Access (OFDMA). For mobile communications it would be 802.16e and uses SOFDMA.

3. UMTS which is developed by European Telecommunications Standardization Institute (ETSI). For mobile voice and data UMTS is a good solution. Its multiple access technique is based on Wide-band Code Division Multiple Access (W-CDMA)

4. EDGE allows people to have a technology which is backward compatible with old GSM system while providing higher transmission rates.

5. WiFi is a synonym for WLAN IEEE 802.11 standard. WiFi can be used both in small homes, apartments, and hospitals and in a wider range like cities. For example Sunnyvale, California, provides a city-wide free WiFi

In wireless communication transmission power is different in different modes of operation. Conventionally, we distinguish between idle, sleep, transmission, and reception states. Here, we are interested in two of them, namely, transmission and reception states. Notices that it has been shown that that energy consumption in idle and receive states are almost the same . The only difference between them is the amount of power spent by amplifying the received signal in receiving states.

However, usually this energy is significantly smaller compared to that one required for transmission. Thus, we concentrate on the latter one.

Table 3.8 shows all these transmission powers for different codecs. As one may observe the difference between power consumption of wireless access technologies available today could be as high as two orders of magnitude proving the importance of choosing an appropriate radio interface and transmission technology one-the-fly. Another observation is the importance of choosing codecs. Indeed, different codecs produce their outputs in a wide range of data rates. Particularly, as we already observed the raw data rate ranges from 5:3Kbps for G.723.1 to 64kbps for G.722 or G.711.

4. VIDEO CODECS

The process of encoding, transmission, decoding, and finally playing back of video sequences is more energy consuming and more complicated than what is discussed in audio codecs. Higher complexity of video codecs imposes more energy consumption in the phases of encoding and decoding than audio codecs. On the other hand, transmission energy in video related applications is mostly more than those applications used for audio transmission. The main focus in this chapter is on these issues and a solution to minimize energy consumption in both phases of coding and transmission and is structured as follow.

Section 4.1 introduces the whole structure of video content communications and compression techniques used in this context and its challenges. The standard used in this thesis is H.264 which is widely used these days. However, this standard has different profiles for different applications and cannot be used interchangeably or might not work properly. These issues are discussed in details in section 4.2 and its subsections. Section 4.3 defines trade-offs for video content communications using H.264 and also provides a method for minimizing energy consumption while keeping the quality of video in an acceptable level.

4.1 Video communications

4.1.1 Monitoring quality of communication

In the context of multimedia communications, there still exist some problems in both well-known best-effort Internet and also cellular networks which affect these services negatively. The reason of these technical problems is inherent features of these communication infrastructures and restrict demands of real-time applications. These requirements of video communications result in their dependency on a minimum level of QoS to be able to perform appropriately. For example, their high sensitivity to losses and also timely delivery constraints must be strictly satisfied.

These issues cannot be handled in best-effort communication infrastructures without collaboration of extra protocols and standards. Using a set of protocols, one can monitor network condition and the application demands so dynamic management of network resources could be possible. One of these procedures for traffic handling is taking advantage of RTP/RTCP while transmitting over UDP.

Real-time Transport Protocol (RTP) is designed to standardize packet structures in multimedia services over IP networks. Additionally, RTP Control Protocol (RTCP) is used in conjunction with it to provide monitoring capabilities as well. RTP defines the format of packets and these packets are sent through IP networks and then RTCP is used to monitor network performance and its QoS.

In order to deal with mentioned problems of real-time data transmission over best-effort communication architectures, RTP packets consist of some additional flags and elements. Timestamps are used for synchronization purposes. To provide the possibility of detecting packet losses occurred during transmission, sequence numbers are also included into these packets. Format of the data which is being carried by the packet presents in each packet so at the receiver side, appropriate process can be executed. After a number of transmission and in a periodic form, RTCP exchanges controlling information between sides of communication so that monitoring and handling traffic of network will be possible.

Monitoring process must be done on all entities of a network involved in the specific session so that each one can dynamically handle possible impairments. For this purpose, whenever sender gets back network information from receiver(s) it spreads that to all of other entities. This way, each component has an updated version of those parameters which are being monitored. It must be noted that RTP mostly owns 95% of the entire traffic while RTCP bandwidth usage is limited to 5% of the whole traffic.

4.1.2 Compression challenges

Compression algorithms are used to reduce bandwidth requirements of media applications. However, at the same time compression brings a number of additional problems to network designers. First of all, compression makes media vulnerable to network impairments such as packet losses and delays. Secondly, modern compression algorithms, especially those, used for video information, are relatively complex involving multiple stages with complicated computational algorithms. It is usually the case that the higher the compression ratio of the algorithm the more complex the algorithm is. Further, compression affects the perceived quality both directly and indirectly. Considering video information as an example, the compression process removes spatial, temporal and psychovisual redundancies encoding the most important parts. It implies that the amount of actual information per data unit in a new compressed bitstream increases.

Although downloading a whole video information like an entire MPEG-4 movie has been a robust way of using the Internet for multimedia applications, streaming is becoming more and more popular these days. Conventional downloading approach demands a high buffer space and user must wait for the information to be downloaded entirely which may take from some minutes to even hours.

Streaming, on the other hand, solved these kind of problems and is being used by many users as it is more flexible than downloading. In this approach, instead of waiting for the whole sequence to be received, each video stream is divided into chunks. These chunks are called packets and then these packets would be encoded at the sender side and transmitted over the network. At the receiver side, these received packets would be decoded and then can be played back.

The process of sending packets at the sender would be on-going simultaneously while the receiver is decoding and streaming already received parts of the video. Although this approach requires less buffer space than downloading, many issues are also introduced. For example the process of encoding, transmitting, decoding, and playing-back that must be done all together would result in a kind of delay problem called low delay. An example of low delay problem can be seen in IPTV applications where user wants to change between channels at any time and this amount of delay must be minimized.

One can distinguish between multimedia communications in a number of ways. One of these classifications takes into account the number of senders and receivers and its components are described as below:

1. Unicast in which one sender and one receiver are connected like media on demand applications

2. Multicast in which there is one sender and a number of receivers. Two types of multicast communications are IP multicast and P2P multicast which is a application level multicast. In the context of bandwidth utilization, multicasting is much more efficient than multiple unicasts.

3. Broadcast in which sender will find all the receivers to them it is connected and then transmits to them.

Communication channels do not have a steady state and because of many things their conditions vary during a multimedia session. That is why, having the ability to monitor channel to see its impairments or improvements is always a plus. This can be done only in unicast communications where one sender and one receiver are connected. In addition to monitoring channel, this feed back makes it possible for the receiver to tell the sender about end-device characteristics as well.

All these information helps sender to decide which codec or codec's profile is the best choice accordingly. Sender has the possibility to adapt the compression ratio, data rate generation, error correction algorithm and other features to optimize the quality of the video received and seen by receiver. This will directly affect QoE and power consumption.

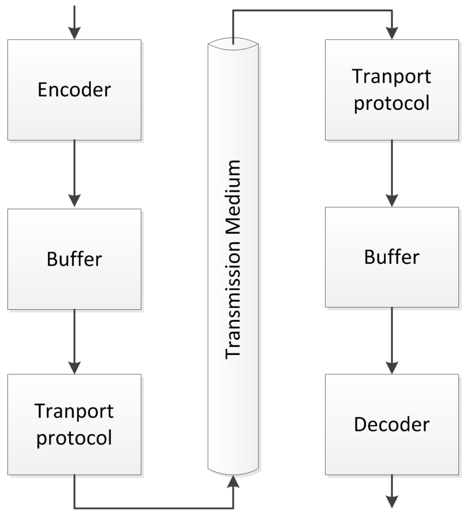


Figure 4.1: Video communication system.

Communicating video content multimedia consists of three main parts each of which includes one or more components. These parts are sender side, transmission medium, and the receiver side as shown in Figure 4.1.

4.2 H.264

In the context of media data, terms codec and container are confusing for many people. A compressor-decompressor or in its shortened form codec, is responsible for either compressing raw media data (which then can be transmitted or stored somewhere) or decompress them (to be viewed at the receiver side after transmission or to be transcoded). Containers are used to hold various media formats. Since different codecs are used for encoding (compressing) raw media data, containers must be able to work with different coding schemes. The more coding schemes a container can handle, a better one it is.

Containers are used in different applications. At this point we introduce a number of most common containers which are wellknown. Advanced Systems Format (ASF) is a Microsoft-based container file which can have extensions like .wmv, .asf, and .wma. The latter, for example, is compressed by Windows Media Audio codec but is an ASF container file. Audio Video Interleave (AVI) is another wellknown codec which is defined by Microsoft. Motion Pictures Expert Group introduced widely used MP4 container. H.264 codec is used for compressing video data and AAC codec for audio data inside MP4 files. Other commonly used containers are Flash, Matroska (.mkv extensions) and DivX.

Two main standardization organizations work in video codecs field which define and provide different standards and protocols. ITU organization which is already introduced in this study and all H.26x standards are defined by ITU. The Internal

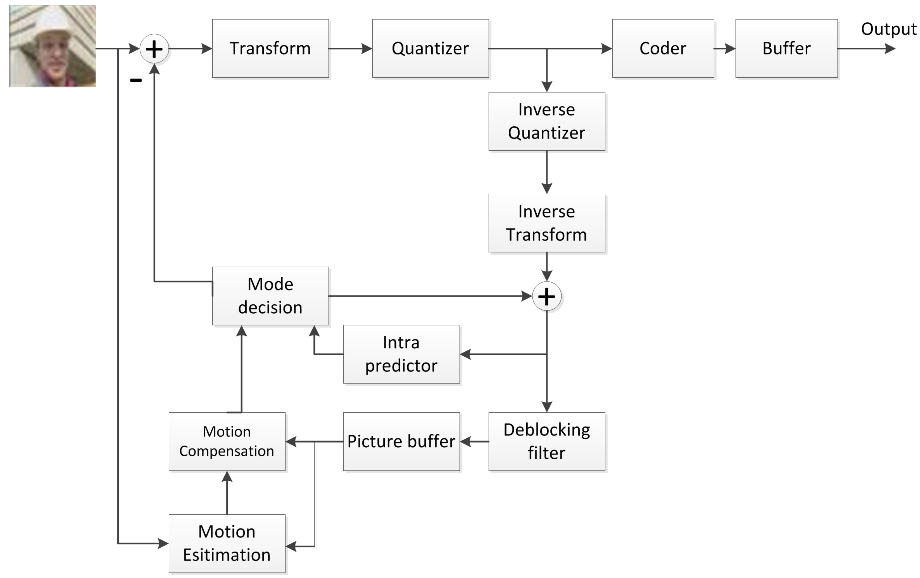


Figure 4.2: H.264 structure, encoder side.

Standards Organization (ISO) is another standardization body which has introduced well-known JPEG and MPEG standards. JPEG is used for still images compression and MPEG is used for moving pictures. This thesis, however, considers most recent and powerful codec. H.264 standard is the latest generation of video codecs which is nowadays widely accepted and used in different applications from streaming to recording [19], [20], [21].

4.2.1 Codec Structure

H.264 improves compression efficiency by reducing data rate around 2 times versus Mpeg-2 and MPEG-4 simple profile. This huge improvement in compression and data rate, makes many on demand applications available over the Internet a well noticeable reduction in the memory space required to store an HD movie.

Encoder

In order to have a better view of this codec, it is necessary to take a look at the structure of the codec. For this reason, Encoder and decoder parts of the codec are separated and the encoder part is depicted in Figure 4.2. Different components of this block diagram are explained below.

1. **Transform** used in H.264 is a 4*4 integer transform with specified coefficients make it possible to be invertible. In this coding algorithm, values of different pixels in both cases of Inter Macro Blocks (MB) or intra MBs are predicted. In the case of Intra MB these values would be predicted from neighboring pixels

of the same picture while in intra mode, previously decoded reference pictures would be used for this purpose.

2. **Quantizer** which is also called scaling is the next block after transformation. Based on the quantization parameter of the intended MB, scaling factor would be chosen. Each MB has a number of sub-blocks within which the element to be scaled can be found. The position of this element also affects the scaling factor value. It also might be possible to use a feed back from Buffer block to Quantizer to enable rate controlling procedure. In that case, the algorithm which is used for rate controlling purpose chooses the quantization parameter.

3. **Coder** After first step, transformed coefficients would be quantized at the quantizer. After all these, there is a need to a block to code these transformed coefficients which happens at the coder block. Two different coding schemes are used at this block which are Context-Adaptive Variable-Length Coding (CAVLC) and Context-Adaptive Binary Arithmetic Coding (CABAC) both of which are which is a lossless compression algorithm. Both of these coding methods are supported in main profile of H.264.

4. **Inversing blocks** The output of Quantizer would be divided into two branches. One of them would be passed to the entropy encoder which then will be buffered and transmitted over channel. The second branch,must go through an inverse transformer to derive (after inverse quantization) primary data. That is why being perfectly invertible is important in transformation process.

5. **Mode decision** It is necessary to choose coding must be done in inter or intra mode which is why this block is placed. According to the rate and distortion and also the algorithm used in rate control, the most efficient coding mode would be selected at this stage.

6. **Deblocking filter** One feature of H.264 for is its possibility of using multiple reference pictures. The task of this filter is providing those references for the pictures of a sequence which are intended to be used as reference pictures. This filter operates on both MBs coded in inter mode or intra mode. In case of inter mode coding, it will be applied to the picture after motion compensation part. For Intra mode coding, it will be applied after intra mode prediction.

7. **Other blocks** Intra prediction block is responsible for predicting intra MBs as it is clear from its name. However, this process is a complex one which is beyond this project. Motion estimation and motion compensation blocks, as it can be understood from their names, are used for predicting inter mode coding scheme.

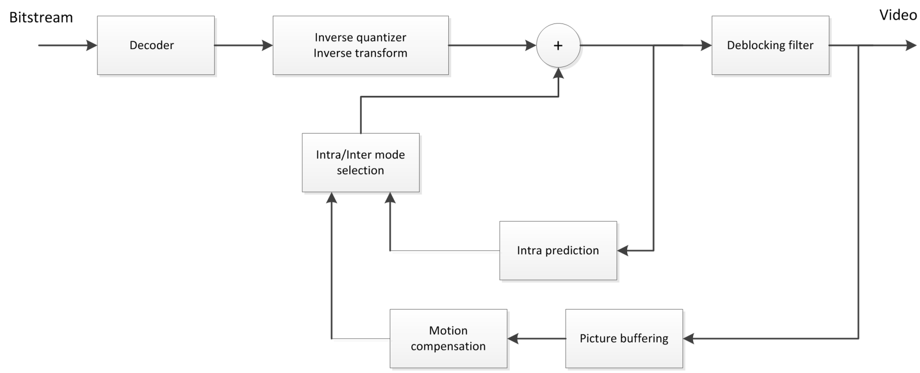


Figure 4.3: H.264 structure, decoder side.

Encoding overview

In encoding process, encoder must distinguish between header and MB elements. These two parts would be encoded using different coding algorithms in H.264. MB level coding, as it is mentioned previously, the first thing that encoder must decide is the entropy coding type to be in intra or inter mode.

Then, According to the macroblock coding scheme which is decided, encoder must be able to perform rate controlling techniques. The reason why this functionality is important relates back to its significant effect on the output quality. In other words, the target of this part is keeping coding bit-rate in the range of maximum bit-rate while keeping the quality of video in an acceptable condition.

Slicing elements of the input video sequence and coding each element using appropriate algorithm and also performing bit allocation on MBs which at this step are partitioned into different blocks, it's time to reconstruct MBs again. At this step MBs are divided into two groups labeled as codec or not coded. If it's not coded it will directly be passed to prediction block to be reconstructed. If Mb is labeled as coded, it must first go through inversing blocks to become like a not coded one and then be passed to prediction blocks.

Decoder

The block diagram of decoder can be seen in Figure 4.3. Components of decoder are the same as what we saw in encoder structure. First block is entropy decoder which distinguishes between coded MBs in CAVLC or CABAC and decodes residual signal in a n appropriate way.

After entropy decoding process, inverse blocks operate on the decoded MBs and according to the mode of coding which can be either intra on inter, the residual block can be reconstructed. This will be done for all blocks of a MB and then for all macroblocks of the picture.

4.2.2 Profiles and levels

It is already mentioned that H.264 is one of the most widely used standards these days meanwhile it is a very complex standard and using it appropriately requires a good knowledge of its features and the requirements and limitations of the specific application for which this standard is intended to be used. Two main parameters of the codec are profiles and levels each of which address different problems.

Profiles

Profiles of H.264 can be used to control different complexities and power consumption due to processing. Different profiles use different encoding techniques for compression. In addition to complexity and power, another trade-offs of these profiles are quality enhancement and the complexity of decoder. More complex profiles take advantage of more advanced compression algorithms which require powerful decoders to be able to decode such encoded sequences.

Number of profiles which are provided by this standard is almost a couple of tens. In each of them, encoder is allowed to use a set of parameters and not allowed to use others depending on what is defined by the profile. Out of these profiles, at this section, we focus on those that are suitable for Scalable Video Coding (SVC) and those that are intended for 2D video compressions. For 3D applications, an extension is introduced named Multi-View Video Coding (MVC) which is not defined in this study.

1. Scalable Baseline Profile (SBP) is intended for applications in which the complexity of decoding must be low. Most of surveillance applications and mobile broadcast are in these group of applications.

2. Scalable Constrained Baseline Profile (SCBP) restrictions of SBP mostly still exist in this profile. However, the decoding complexity is a bit more than SBP and in real-time applications, this profile would be used.

3. Scalable High Profile (SHP) For streaming and storage purposes, this profile is an appropriate choice. Restrictions of SBP and SCBP do not exist here and coding with arbitrary resolution ratios is supported.

4. Scalable High Intra Profile (SHIP) is used where ever professional applications using H.264 are targeted.

Levels

Levels make it possible to manage maximum data rate that can be served in an acceptable range of quality on the link. In other words, it is used for bandwidth utilization controlling purposes. More over, different levels provide options to control maximum resolution and solve possible memory issues which might happen at the decoder side. To clarify, lower levels mean that the maximum bit rate and resolution must be below or equal to a specific level. The idea of controlling max resolution is to make it possible for devices with different capabilities to be able to decode and play back H.264 encoded bitstream.

For example, assume that baseline profile is chosen for encoding input bitstream which imposes its encoding rules on the encoding parameters. Then, using levels, one can guarantee that the picture resolution would fit the targeted resolution and the bit rate and frame rate would be in a range that is acceptable for a specific device. It must be mentioned that, while a level is chosen, the decoder must be capable of handling all levels below thatl for the defined profile.

4.3 Trade-offs in H.264

This study investigates trade-offs on wireless networks for which multimedia is one of the most important applications. Transmission of voice over IP networks and its challenges and the trade-offs has been presented in Chapter 3. However, in case of video content, it is more complex to compare compression, QoE, and energy efficiency.

H.264 standard has been studied in this paper for which among all profile introduced in previous section, only baseline and constrained baseline profiles can be used for wireless multimedia communications. However, it is not possible to simply change between these profiles and calculate the bit rate and compression ratio for video codecs. The reason is that, the same profile at the same level of operation for two different video sequences might show totally unequal results.

On one hand, video slides have various motion conditions. On the other hand, encoding each video sequence highly depends on these movements of the video. The reason relates back to the coding schemes which use different coding techniques for motion compensation. As a result, encoding a video sequence containing 100 frames of a smooth nature scene requires much less encoding time and energy than coding a high movement sequence of 100 frames of a football match.

However, it is necessary to find an approach to compare three above-mentioned components on video codecs as well. The reason is that encoding video sequences require much more energy and time than voice data. More over, encoding video bitstreams result in high data rates and their transmission also impose a high power consumption cost. Choosing an inappropriate codec might result in either a very low battery lifetime duration which is not acceptable for users or an acceptable battery usage but poor quality.

In the following sections, at first the method used is described and then according to the methodology different components of tradeoff are compared to each other.

4.3.1 Methodology

To analyze the effect of encoding parameters on bit rate, complexity, and power consumption in the context of video sequences, two issues must be considered and solved. Firstly, the parameters must be chosen in a way that encoding and decoding processes be applicable for real-time applications. For this reason, data rate of the output must be tenable for these applications. Complexity of coding algorithm is another issue which might be so that portable devices processors be able to do handle it. Power consumption must be also acceptable for batteries of these devices.

Since H.264 is a widely used standard these days, it is better to see tradeoffs on its codecs rather than other old algorithms. However, as it is mentioned previously only baseline profile can be used and there is a need to see different levels of baseline profile effect on intended parameters.Another main problem is that it is not possible to use different sequences and encode them using available codecs and compare them with each other as motion of sequences has a great impact on the result. To solve this, the scope of this project is only on sequences with low/medium motion conditions.

To overcome the problem of profiles which can be used, baseline profile of H.264 can divided into 9 different complexity levels. Each of these levels are called Complexity Parameters (CP) which is similar to what is done by [23]. Each complexity parameter corresponds to a certain amount of processing level that are separated by the coding algorithms used in them. These differences can be seen in the searching techniques they use, prediction schemes and other factors which can strictly affect both efficiency and complexity of encoder.

Using this approach, makes it possible to compare power consumption of different practical codecs. However, these CPs do not affect data rate, PSNR, and SSIM that are important factors. In other words, 9 different codecs which are all suitable for real-time applications and consume different amount of energy are in hand. However, the tradeoff between QoE and compression is not achieved yet.

Calculating encoding energy consumption for different complexity parameters of H.264 baseline profile, enable us to calculate PSNR and bitrates. For this purpose, this study takes advantage of different quantization steps named Quantization Parameter (QP). for different QP values and different sequences. The most important assumption that is used is that for a given QP value, the bitrate and PSNR will almost remain the same.

In other words, as it can be seen in [23] and [24], data rate for the CP values between 2 to 10 for a given QP do not change noticeably. The change in PSNR is less than 1 dB in total and the change in bit rate is almost 1 kbps. In addition to

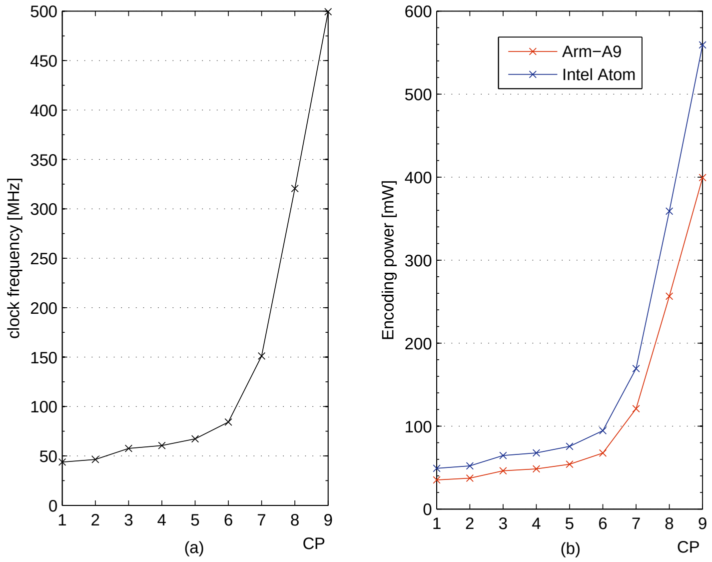


Figure 4.4: Different CPs within baseline profile frequency clock and encoding power.

that, within a single CP changing QP does not effect cycles per second (MHz) of CPU clock which is used to calculate processing power.

This paper, uses these assumptions and scales different CPs from 1 to 9. In this range, both data rate and quality metrics (PSNR and SSIM) can be assumed constant. There are two things that affect data rate i.e. different QP values and also different sequences. The latter is because of the motion of different sequences. In this paper four different sequences with a medium to low motion are considered to minimize the impact of motion of sequences.

4.3.2 Energy required for compression and transmission

Processing power

This subsection investigates different aspects of energy requirements using introduced method. At first, processing power of baseline profile for each of those CPs are calculated. For transmission power, the same algorithm used in audio codecs will be applied. I.e. having data rate for different QP values of a specific CP makes it possible to estimates power consumption for transmission over different technologies.

Figure 4.4.a maps different complexity levels to their corresponding clock frequency in MHz. This figure presents a monotonic ascending schemes in clock frequencies of different levels of CP.Then using this frequencies, it is possible to calculate encoding power consumption for different processors.

To calculate encoding power first one must find an appropriate processor. In

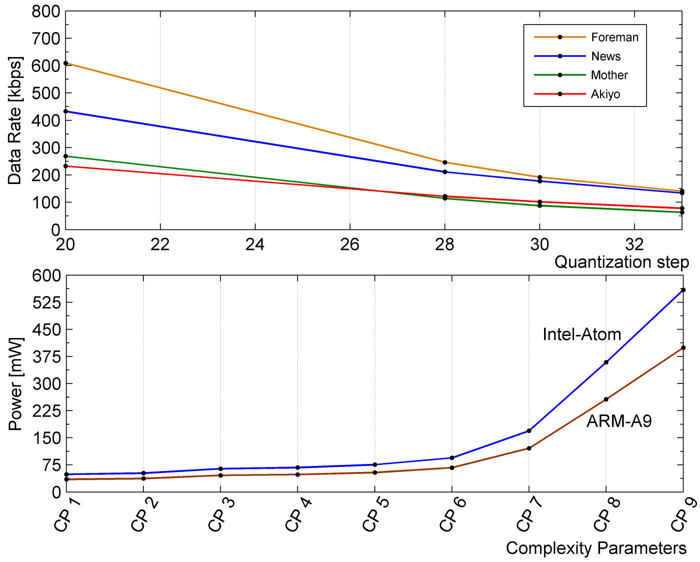


Figure 4.5: Power consumed in compression in different data rates.

this study two newly released processors which are used in many smart phones and tablets have been investigated. Intel atom with x86-64 instruction set and ARM Cortex-A9 with instruction set ARMv7 are chosen for this purpose. It must be noted that they will be used in dual-core or quad-core mode on powerful devices.

According to [35] Intel dual-core processor consumes 1.12 mW per MHz and ARM Cortex-A9 consumes 0.8 mW. Knowing this amount of energy consumed per cycles per second and also Figure 4.4.a one can calculate processing power consumption for different CPs using these two processors. Figure 4.4.b presents this. As it can be seen Intel processor consumes more energy than ARM.

After all, to see the efficiency of different CPs in the context of compression and processing power consumption, it is required to consider Figure 4.5. This figure presents the trade-off between data rate of the output of encoder and the power that encoder consumes to encode in that datarate. Both processors are again compared in this figure.

Transmission

As it is already mentioned, the factor which affects transmission energy is QP as changing QP result in variation of data rates. The trend of transmission power change has a direct relation with QP. Increasing QP will reduce data rate which result in decreasing the amount of energy required for transmission.

To see the effect of QP on transmission power better, it is required to see its influence on different sequences. For this reason, Four different YUV sequences are investigated here. For each sequence the impact of 4 different QPs of 20, 28, 30, 33 are examined and the transmission power over different transmission technologies is calculated.

Table 4.1: Transmission energy for different sequences [mW].

(a) Foreman

QP	LTESISO	LTECLSM	WiMAX	UMTS	Edge	WiFi TTI=300
20	15835	5481.2	974.4	6090.2	522.7	101.5
28	6405	2217.1	394.1	2463.4	211.4	41
30	4986	1725.8	306.8	1917.6	164.6	31.9
33	3659	1266.7	225.2	14.07.4	120.8	23.4

(b) News

QP	LTESISO	LTECLSM	WiMAX	UMTS	Edge	WiFi TTI=300
20	11256	3896.2	692.6	4329.1	371.6	72.1
28	5489	1900	337.8	2111	181.2	35.2
30	4613	1596.7	283.9	1774	152.3	29.6
33	3482	1205	214.27	1339	114.9	22.3

(c) Mother and Daughter

QP	LTESISO	LTECLSM	WiMAX	UMTS	Edge	WiFi TTI=300
20	6986.2	2418.3	429.9	2687	230.6	44.8
28	2969	1027.7	182.7	1419	98	19
30	2290.1	792.7	140.9	880.8	75.6	14.7
33	1645.8	569.7	101.3	633	54.3	10.5

(d) Akiyo

QP	LTESISO	LTECLSM	WiMAX	UMTS	Edge	WiFi TTI=300
20	6042.9	2091.8	371.9	2324.2	199.5	38.7
28	3166.3	1096	194.9	127.9	104.5	20.3
30	2647.1	916.3	162.9	1018.1	87.4	17
33	2034	704.2	125.2	782.4	67.1	13

Table 4.1 presents calculated values of these parameters. As it can be seen in the four tables provided for different sequences, at QP=20 the highest amount of energy is required for transmission while QP=33 requires least amount of energy as expected. The reason relates back to the data rate. However, impact of higher QPs on QoE which would be examined in the following sections must not be neglected.

Another important factor which can be proven by Table 4.1 is the possibility of trading-off power consumption of encoding and transmission. This issue was introduced in audio codecs as well. However, in that case, the energy for transmission was mostly more than processing energy as compression of voice is not that complex.

This condition is totally different considering video codecs. Most of the energy required for processing procedure of video codecs is in encoding phase.

4.3.3 PSNR and SSIM

Nowadays, the subjective MOS-based methods proposed by ITU-T are the only standardized methods for assessing quality of video information. These tests involve an audience of people watching a video sequence and deciding on its quality under specific environmental conditions. Statistical analysis of the collected data is used to evaluate the quality of user experience. Evaluating the MOS for video information is more complicated compared to voice information and usually require more respondents to provide a reliable estimate. Although some progress have been made over the past years, objective performance metrics that correlate well with MOS grades and would automate the process of video quality assessment are still under development. Standardization of such metrics is the main task of video quality experts group (VQEG) that was founded in 1997.

Based on the amount of information about the original video available at the point of evaluation objective quality assessment algorithms are classified into full-reference (FR), reduced-reference (RR), and no-reference (NR). FR metrics require the entire frame sequence to be available at the point of evaluation. Due to unavailability of the video sequence at the receiving end FR algorithms cannot be used in networking environment. NR metrics deal with distorted video only and do not require the reference sequence at all. The main problem of NR metrics is that it is often impossible to distinguish between compression/network induced distortions and specific features of a particular content. Thus, the design of NR algorithms is extremely complicated task and only limited progress has been made so far implying that on-the-fly quality control based on perceived quality assessment is not possible nowadays. RR metrics are a compromise between FR and NR metrics. They extract a limited set of features from the reference video and use this information at the point of evaluation. This allows to distinguish between compression/network induced distortions and specific peculiarities of a video sequence. RR algorithms can be used in network environment given that some information about the original signal is transmitted to the receiver.

The goal of any objective QoE metric is to produce video grading that correlate well with grades produced by MOS tests. There are a number of approaches to address this problem. One of those is to capture properties of the human visual system (HVS). However, operation of HVS is still poorly understood making it impossible to isolate those properties that influences the perceived quality. During the first call for proposals issued by VQEG a number of HVS-based algorithms have been submitted. Detailed evaluation of these algorithms demonstrated that none of those performed better compared to conventional MSE/PSNR metrics. Additionally, HVS-based algorithms are always of FR type restraining their application in network environment. Indeed, even partial information about the original signal is rarely available at the receiving end.

Both MSE and PSNR are easy to compute. Over the years, researchers make themselves familiar with MSE/PSNR allowing them to interpret the values immediately. The main reason against using MSE/PSNR quality metric is that it was reported to have poor correlation with MOS grades. However, studies of VQEG group demonstrated that MSE/PSNR provides performance comparable to advanced FR metrics based on capturing properties of HVS system.

Another metric that is used in this work is structural similarity index (SSIM). The idea behind SSIM is that images are highly structured, and that the eye is mostly sensitive to structural distortion. The SSIM index expresses perceived quality by comparing local correlations in luminance, contrast, and structure between reference and distorted images. Performance of SSIM index was reported to be significantly better than that of MSE/PSNR metric. The algorithm is also simple and can be easily implemented in software. SSIM was used as a basis for video quality assessment algorithm is several studies

QoE after compression

Lossy compression is one of two major reasons for quality degradation of video. To evaluate the perceived quality after compression for a wide range of parameters we encoded original YUV video sequences and then estimated PSNR and SSIM at the output of the codec. For encoding purposes a reference JM v.15 H.264 has been used. In all encoding procedures the frame per second (fps) was set to 30 and 102 frames were chosen from the YUV sequences to be encoded in QCIF format. We repeated the process for all four sequences and all QP values we have chosen previously.

Simulation results of encoding process for PSNR, and SSIM was as expected. Table 4.2 presents the result obtained from these simulations for all four sequences and QP values. The trend of decrement in PSNR, SSIM, and bitrate by increasing QP, is presented in the table. The highest SSIM obtained in all cases is equal to 0.98 out of 1 in the least QP equal to 20. As it is already mentioned, sequences motion are chosen to be in an acceptable range of similarity with each other. As a result, it is convenient to compare these values with each other. Bitrate is given in kbps and PSNR is in dB.

Although it is always better to consider all factors that affect QoE, calculations of PSNR and SSIM which are presented in Table 4.2 are important as they show the efficiency of encoder of the codec. Decoder side of the codec has some options for concealing errors like packet losses which are not enabled in this specific part of

Table 4.2: Bitrate, PSNR, and SSIM after compression of H.264 different complexity parameters.

Sequences	Parameters	20	28	30	33
Foreman	Bitrate	609.02	246.34	191.76	140.74
	PSNR	42.56	36.89	35.53	33.57
	SSIM	0.98	0.95	0.94	0.93
News	Bitrate	432.91	211.1	177.41	133.92
	PSNR	43.04	37.24	35.61	33.47
	SSIM	0.98	0.96	0.96	0.94
Mother and Daughter	Bitrate	268.7	114.19	88.08	63.3
	PSNR	43.45	37.98	36.63	34.78
	SSIM	0.98	0.95	0.91	0.91
Akiyo	Bitrate	232.42	121.78	101.81	78.24
	PSNR	44.38	38.78	37.3	35.25
	SSIM	0.98	0.97	0.96	0.94

experiment. The reason is that it is desired to see how a codec would operate when channel is ideal.

In this condition, original sequence would be compared to the decoder output and encoder reconstruction output. The latter is produced in encoding process and used while decoding steps as reference pictures. Number of references in all parts of this study is considered to be 1.

QoE after transmission

A transmission medium is the second most important source of quality degradation. In this subsection we investigate the effect of packet losses on the perceived quality by arti

cially introducing packet losses into compressed stream of packets. These packet losses are an inherent characteristic of any wireless access technology of today and in real life applications, users face them repeatedly while watching an online television or videoconferencing. Degradation range due to this issue might change between "negligible" to "unacceptable".

This range not only varies as a function of Packet Loss Rate (PLR) but the pattern of packet losses affect video quality as well. In this paper we assume that in spite of various error correction procedures implemented at the data-link layer of modern wireless access technologies there are some incorrectly received symbols left unrecovered resulting in loss of IP packets. Secondly, we assume that the packet loss process is memoryless. Below we explain why these assumptions do hold.

AMC system

Generic AMC system works as follows. The range of the received SNR is divided into a number of intervals. These intervals are chosen such that a certain combination of the constellation scheme and the coding rate provide the best possible spectral efficiency in each range of SNR. CSI in terms of the received SNR is fed back to the transmitter on a frame-by-frame basis. When SNR changes, new constellation scheme and coding rate are chosen and further used. AMC could provide significant gains in terms of optimal channel usage often measured in terms of spectral efficiency. However, its performance heavily depends on accuracy of SNR estimation at the receiver and timely delivery of this information to the transmitter. Therefore, to effectively use AMC system the channel fading process must be slower than the SNR feedback sent from the receiver. Nowadays, AMC is used in many modern wireless access technologies including UMTS, IEEE 802.16, etc. Combined with MIMO, AMC may provide good results in very complicated propagation environments.

Observe that AMC system is, in fact, a performance control system optimizing spectral efficiency of wireless channels for a given frame error rate (FER) using both FEC and modulation based on SNR feedback. Here, FER is the performance metric of interest. In other words, AMC tries to keeps FER at a certain pre-de

ned level while changing rate of a channel. It is important to note that we may in principle set FER to any given value getting different rate for each modulation.

HARQ system

AMC does not completely eliminate incorrectly received packets. In some wireless access technologies ARQ system is used on top of AMC. ARQ acts reactively by retransmitting incorrectly received frames. There are a number of different variants how ARQ is implemented. Nowadays, selective-repeat ARQ (SR-ARQ) retransmitting only incorrectly received frames are widely used. The most important performance parameter is the number of retransmission attempts, r. When r is large the effective FER is virtually zero. Usually, r is set to small finite number there is some residual packet loss probability. Thus, even when AMC and ARQ are both used there the packet loss probability may still be non-negligible.

Interleaving and OFDM

The received signal strength process of a wireless channel is known to have a strong memory. Consequently in most performance evaluation studies bit, frame, and packet loss processes are considered to have some memory. While it may indeed by true for the bit error process as some memory is in fact retained. However, modern techniques such as OFDM and interleaving make frame and packet error process almost memoryless. Although interleaving and OFDM are two completely different having different aims they both affect the bit error process in such a way that it losses memory. According to OFDM a whole bandwidth is divided into subchannels and user information.

Recall that OFDM was originally proposed to combat inter-symbol interference

(ISI) by reducing the symbol rate of a channel. OFDM is implemented by dividing the whole channel into a number of subchannels with much lower information rates. The channel symbols transmitted by a mobile are spread over these channels. Since the original channel is often frequency-selective (i.e. different frequencies experience different attenuations and phase shifts) it is unlikely that channels carrying successive symbols would fade significantly reducing dependence between results of consecutive symbol transmissions. As opposed to OFDM reducing memory is the original aim of interleaving. Taking into account that most wireless access technologies are currently based on OFDM channel organization and all of them implement interleaving the packet loss process is almost memoryless.

The experiments were performed as follows. First, we encoded the YUV sequence in RTP mode using JM v.15 implementation of H.264 codec. Let N denote the number of RTP packets obtained. To emulate the effect of lossy transmission medium a sequence of 0 and 1 of length N - 3 is then generated, where 0 denotes a correctly received packet while 1 corresponds to the lost one. As discussed in Section 2 packet losses we assumed to happen in uncorrelated manner with a certain packet loss probability (equivalently, packet loss ration, PLR for short), i.e. completely randomly implying that there is no intentional grouping of losses. This sequence was further applied to the packetized video stream starting from the 4th packet by dropping and keeping packets. Note that we explicitly assumed that first three packets of all sequences are always received correctly. The reason is that these packets carry important information related to the choice of decoding procedure. Loosing at least one of them would fail the decoding process completely. Although for the tested sequences it is rather signi

cant potion of the content (due to short playback time) in real-life applications the consequences of this assumption are almost unnoticeable. Finally, after encoding and dropping RTP packets we decode the compressed bitstream. A decoder side of H.264 codec has two options for concealing errors like packet losses. Frame copy and motion copy are available algorithms in JM v15.

There are a number of hidden issues one have to deal with doing such experiments. For example, in this study we need to compare the effect of the packet loss rate on sequences encoded with different QP. As we already saw changing QP produce different compressed data rate. Since the RTP packet size depends on the amount of information produced by a codec per frame generation interval (1=30 for 30fps video) loss of a single packet would affect different amount of compressed data for different QP. One way around this problem is to measure the loss rate not in packets but in some rather small reference units, e.g. bytes. Further, it is also possible to add the effect of grouping of packet losses by introducing dependency into sequence of 0s and 1s. For, example, discrete autoregressive process of order one, DAR(1), provide an easy way to generate sequences of 0 and 1 having some predefined lag-1 autocorrelation and packet loss rate.

Furthermore, the selection of the loss concealment algorithm is another issues that has to be addressed. Most real life implementations do not use advanced error concealment schemes relying solely on replacing the lost frame with the previous one that has been decoded correctly (frame copy). Motion copy is another simple way to replace regions of a frame contained in the lost packet. Those advanced techniques based on interpolation brings additional complexity of implementation that should better be avoided in hand-held devices. Finally, error concealment techniques rarely provide significant performance gains. Finally, during decoding process we observed that for PLRs more than 12 percent decoder could not operate correctly and the decoding process eventually stops with error.

Table 4.3: PSNR and SSIM under different PLRs and QP cases shown in C_{PLR}^{QP}.

(a) QP=20,28

Seq.	**Parameters**	$\mathbf{C}_0^{20}$	$\mathbf{C}_3^{20}$	$\mathbf{C}_5^{20}$	$\mathbf{C}_{10}^{20}$	$\mathbf{C}_0^{28}$	$\mathbf{C}_3^{28}$	$\mathbf{C}_5^{28}$	$\mathbf{C}_{10}^{28}$
Foreman	PSNR	42.7	29.67	28.77	27.57	36.69	29.39	28.47	26.66
	SSIM	0.98	0.89	0.78	0.75	0.95	0.9	0.79	0.61
News	PSNR	43.04	37.39	29.73	30.03	37.24	33.8	37.07	28.95
	SSIM	0.98	0.9	0.81	0.72	0.96	0.89	0.79	0.72
Mother and	PSNR	43.57	37.64	36.44	36.91	37.98	35.24	34.45	34.53
Daughter	SSIM	0.98	0.87	0.81	0.75	0.95	0.9	0.86	0.8
Akiyo	PSNR	44.38	40.99	37.47	37.3	38.78	36.91	34.89	35.03
	SSIM	0.98	0.94	0.85	0.82	0.97	0.91	0.83	0.79

(b) QP=30,33

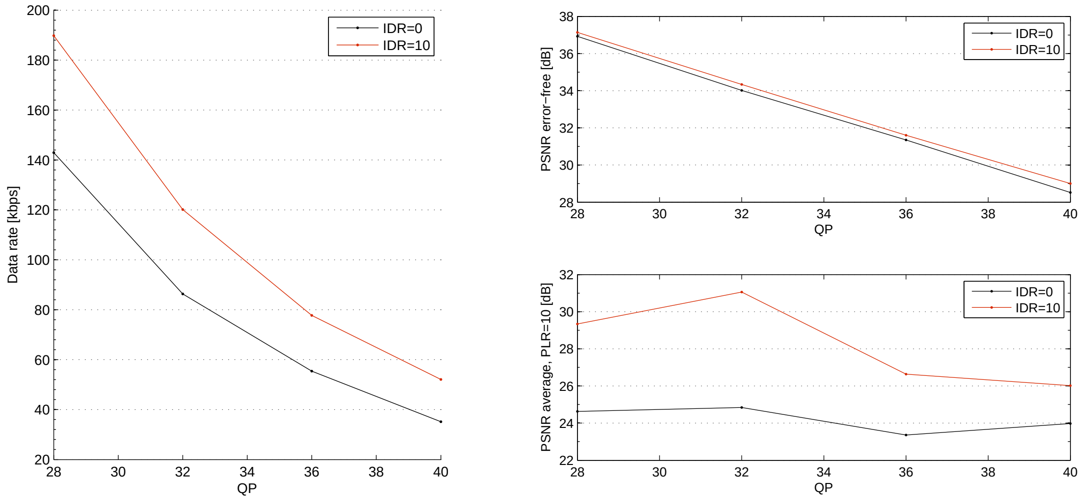
Seq.	**Parameters**	$\mathbf{C}_0^{30}$	$\mathbf{C}_3^{30}$	$\mathbf{C}_5^{30}$	$\mathbf{C}_{10}^{30}$	$\mathbf{C}_0^{33}$	$\mathbf{C}_3^{33}$	$\mathbf{C}_5^{33}$	$\mathbf{C}_{10}^{33}$
Foreman	PSNR	35.18	29.19	28.17	26.26	33.17	28.58	27.7	25.89
	SSIM	0.94	0.89	0.78	0.62	0.93	0.87	0.78	0.6
News	PSNR	35.61	32.61	28.72	28.59	33.47	31.47	29.07	28.29
	SSIM	0.96	0.92	0.82	0.79	0.94	0.89	0.78	0.71
Mother and	PSNR	36.63	34.42	33.7	34.01	34.78	32.94	32.46	32.43
Daughter	SSIM	0.91	0.89	0.8	0.76	0.91	0.88	0.85	0.76
Akiyo	PSNR	37.3	35.67	34.12	34.0	35.25	33.92	33.11	32.75
	SSIM	0.96	0.91	0.85	0.79	0.94	0.87	0.79	0.78

Table 4.3 summarizes PSNR and SSIM values of different sequences under packet loss conditions. The general trend is similar to what we expected, i.e. both metrics decreases when PLR increases. One should also observe that the degradation is more significant for sequences with moderate motion (Foreman, News) compared to those having low motion (Mother and Daughter, Akiyo).

Instantaneous decoding refresh (IDR) feature of H.264 codec allows to explicitly control the frequency of appearance of intra-coded I frames. These frames serve as a reference points for successive inter-coded frames and serve as a access points for a decoding algorithm. Increasing the IDR frequency may increase resistibility of video to packet losses as it explicitly limits the error propagation distance. All previously provided results are obtained setting IDR to zero implying that there is no pre-defined distance between I frames. In this case the frequency of I frames is dictated by the content of the video only and intra-encoding threshold. Setting it to a certain values almost always results in better perceived quality after encoding and after transmission.

To see the effect of IDR frequency, PSNR of foreman sequence for 4 different QPs have been calculated. We consider PSNR in loss-free environment (equivalent to PSNR after decoding) and with 0:1 packet loss probability. Point estimates of PSNR obtained using five experiments are shown. Fig. 4.6(a) demonstrates the impact of IDR parameter on the compressed data rate. We see that for all values of QP the data rate increases. For this moderate motion video the difference is in fact noticeable, i.e. 34 percent for QP 28 and 57 percent for QP 40. As one may further observe analyzing the results provided in Fig. 4.6(b) this difference does not produce any noticeable effect in terms of PSNR for loss free environment. Indeed, for all values of QP the difference is less than 1dB. In other words, the gain in the perceived quality we get enabling IDR feature of H.264 does not make up for additional power required for transmission. It is important to note that the complexity of compression increases with IDR enabled. However, this increase will not make up for drastically increasing compressed data rate and corresponding energy consumption due to transmission.

The situation is drastically different when packet losses are non-negligible. It could be as large as 16dB (QP 32) implying that setting IDR to a certain (rather small) value can be beneficial for end users in networking environment. Observe that for some QP the values of PSNR corresponding to IDR=10 are close to those of loss-free environment. There are two additional important factors that can be observed in Fig. 4.6(b). First of all, the value of PSNR does not continuously decreases when QP goes from 28 to 40. This effect was expected as PSNR is in fact a logarithmic and PSNR of the sequence after compression with QP=40 is lower than that of sequence with QP=28. The second one is that the difference in PSNR gets smaller as QP increases. To conclude, trading-off energy and QoE is even more complex with IDR enabled. For example, a noticeably better quality at the cost of more transmission power consumption, higher bandwidth requirement, and more processing power can be obtained.



(a) data rate. (b) PSNR.

Figure 4.6: Impact of IDR on data rate and PSNR.

5. CONCLUSION

This thesis examines the trade-offs between quality of experience, compression, and energy consumption. Energy consumption is divided into energy required for transmission over different technologies and energy spent for processing compression algorithms in voice and video codecs.

Results obtained in the thesis prove that implementing a system that modifies factors involved in media streaming under different conditions provides the best possible performance in multimedia real-time applications. For example, changing data rate based on network condition, would improve QoE. Another example is seen in video codecs, where modifying QP makes it possible to trade-off between data rate, QoE, and even energy consumed for transmission.

In addition to the above-mentioned parameters, the compression algorithm used in different voice codecs and in different levels of baseline profile of H.264 video codec, affect both QoE after transmission and power consumption. More complex algorithms increase the compression ratio, which conserves transmission energy. However, packetizing more and more bits and bunding them into one UDP frame (which is the most used protocol in real-time applications) under poor network conditions with high packet loss ratio degrades QoE severely; each lost packet means huge amounts of lost data.

Introduced trade-offs in the project are not constant. For example, new air interface technologies bring different energy consumption values; these affect total calculations of energy requirements, which must be considered in the decision-making process. Improvements in network conditions also make it possible to choose more complex codecs with higher compression ratios without any QoE degradation. The same possible changes in trade-offs happen by introducing new compression techniques.

The possibility of changes in trade-offs, however, does not mean that implementing the system is impossible. Considering all potential changes, such a system can still be implemented and be valuable for certain parameters of multimedia streaming applications in portable devices. These systems provide users with high performing, long-lasting devices.

Comparing energy consumed for transmission and compression, which is an important metric in trade-offs, affect other parameters. This clarifies one reason why implementation of such a system is important. For example, the coding process of foreman sequence in QCIF resolution varies from 35 mW in the lowest complexity mode in a ARM-A9 dual core processor to 559 mW at the complexity level 9 of baseline profile in an Intel Atom dual core processor. Transmission power for the same sequence and in the same resolution varies between 23 mW while the QP is set to 33, resulting in the lowest PSNR considered in this paper, to 15835 mW over LTE SISO setting QP to 20, yielding the best PSNR.

These values for transmission and compression energy consumption proves the importance of minimizing energy consumption while obtaining the best possible quality. However, this thesis considers QoE as a single variable estimated by objective measurement methods, that is just a basic assumption of quality of experience. In order to have a more accurate estimation of QoE, different factors, such as the context of application, the user's characteristics, and more technical influence factors such as the protocol used for transmission, must be considered.

Any change in one factor might improve or degrade another factor,affecting the final QoE parameter. On the other hand, at each instant of time t_i for a specific application, there exists a set of factors that might be replaced with other ones at time t_{i+1}. There could be some cases in which these factors are not replaced, but their influence weight could change. This weight could be the direct effect of a single factor on QoE or the effect of its correlation with some other components.

Considering the potential influence of these metrics on QoE, and examining trade-offs between these components is quite complicated without performing a general but precise study. This paper tried to prepare the background for such a future research project while maintaining accuracy and examining various parameters affecting the trade-offs.

Putting all of these parameters together and viewing the results in one figure would be helpful. However, because of the variety of parameters, this kind of figure might be confusing, specifically in the case of video codecs. Figure 5.1 presents all of these trade-offs covering different audio codecs.

In this figure, the QoE estimator is considered the well-known MOS value. The minimum level of acceptance for QoE, according to MOS grading values, is 3.5. Different codecs are replaced with their corresponding complexities (MIPS) in the x-axis, as the most important feature of each codec in this context is its complexity. The y-axis is divided into two parts. The first part presents values of one to five for QoE, and the second part, which varies between zero to six, presents energy consumption for each complexity level in mW. The DSP considered for encoding is Cortex M3, and the technology over which encoded packets are transmitted is WiFi with TTI value below 15 ms.

According to Figure 5.1, for some complexity levels, which are nominators of one

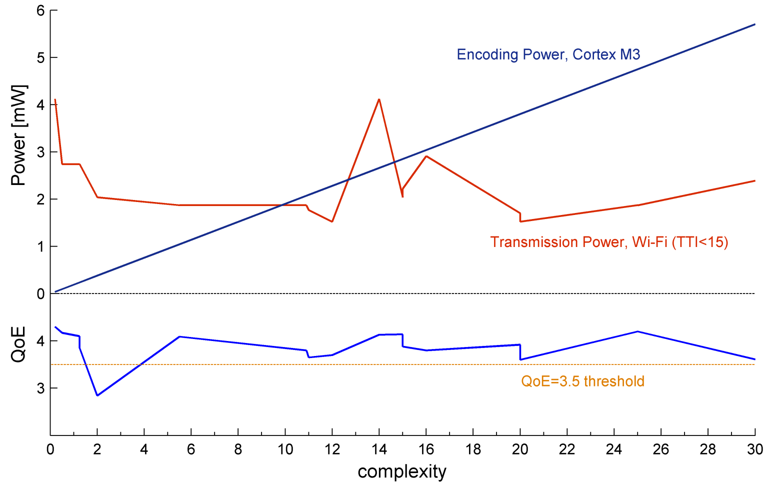


Figure 5.1: Trading-off QoE, compression ratio, and energy efficiency in audio codecs.

or several different audio codecs, obtaining better QoE while maintaining or reducing power consumption might not be feasible. In this context, one might consider the total power as the sum of encoding and transmission powers. At the complexity levels 12 and 14, there is a slight increase in QoE, while the total power consumption shows a noticeable increase. Another example is the codec with complexity level 2; here, the total energy consumption is very low, but the QoE of this codec would not be tolerated in certain real time applications.

An important point raised by this figure is the ability of codecs to operate despite impairment. All calculations were done assuming perfect network conditions (in contrast to video codecs in which different packet loss rates have been studied during the calculation process of PSNR and SSIM). This is one reason why some codecs with higher complexity levels result in lower MOS values compared to some less complex codecs. If the same calculations were performed considering different values of delay, jitter, and packet losses the curve would have revealed different results.

BIBLIOGRAPHY

[1] R. Trestian, A-N. Moldovan, O. Ormond, and G-M. Muntean, Energy Consumption Analysis of Video Streaming to Android Mobile Devices, IEEE/IFIP Network Operations and Management Symposium, (NOMS), April 2012

[2] ITU-T, "Batteries for portable ICT devices," ITU-T TechWatch Alert, Feb. 2010

[3] S-J. Kwon, Y.W. Chung, and D.K. Sung, "Queueing Model of Sleep-Mode Operation in Cellular Digital Packet Data," IEEE transactions on vehicular technology, vol. 52, NO. 4, JULY 2003

[4] G. Bernardi, P. Buneman, and M. K. Marina. Tegola Tiered Mesh Network Testbed in Rural Scotland. In WiNS-DR 08, San Francisco, CA, Sept. 2008

[5] V. Pejovic and E. Belding. Energy-Efcient Communication in Next Generation Rural-Area Wireless Networks. In CoRoNet10, Chicago,IL, September 2010

[6] ITU-T, "Batteries for portable ICT devices," ITU-T TechWatch Alert, Feb. 2010

[7] M.zorzi, R. R. Rao, "Error control and energy consumption in communications for nomadic computing," IEEE transactions on computers, March 1997

[8] R. Trestian, O. Ormond and G-M. Muntean, "On the Impact of Wireless Traffic Location and Access Technology on the Mobile Device Energy Consumption", The 37th IEEE Conference on Local Computer Networks (LCN), October 2012

[9] T. Chen, H. Kim and Y. Yang, Energy Efficiency Metrics for Green Wireless Communications, International Conference on Wireless Communications and Signal Processing (WCSP), pp. 1 6, 2010.

[10] Codecs, A reference guide to all things VOIP, from http://www.voip-info.org, June 17, 2012

[11] A.D. Clark, Modeling the effects of burst packet loss and recency on subjective voice quality, 2nd IP-Telephony Workshop, Columbia University, New York, April 2001.

[12] Rosenbluth, J. H., Testing the Quality of Connections having Time Varying Impairments. Committee contribution T1A1.7/98-031

[13] Alain Hore, D. Ziou "Image quality metrics: PSNR vs. SSIM," IEEE International Conference on Pattern Recognition, 2010

[14] Z. Wang, A. C. Bovik, H. R. Sheikh, and E. P. Simoncelli, Image quality assessment: From error measurement to structural similarity, IEEE Trans. Image Processing, vol. 13, Jan. 2004.

[15] Z.wang, E. P. Simoncelli, and A. C. Bovik, "Multi-Scale StructuralSimilarity For Image Quality Assesment," 37th IEEE Asilomar Conference on Signals, Systems and Computers, Nov. 2003.

[16] B. Pham, O. Wong, "Handheld Devices for Applications Using Dynamic Multimedia Data," Proc. 2nd International Conference on Computer Graphics and Interactive Techniques in Austalasia and South East Asia, Singapore, 123-130, 2004.

[17] BJORK, S., BRETAN, I., DANIELSSON, R., KARLGREN, J. AND FRANZEN, K. (1999). "WEST: A Web Browser for Small Terminals." CHI Letters Vol 1, 1: 187 - 196.

[18] LI, C.-S., MOHAN, R. AND SMITH Multimedia content description in the InfoPyramid. Proceedings of the 1998 IEEE International Conference on Acoustics, Speech, and Signal Processing, 1998

[19] E.Baalkini, H.Sbeiti, S.Niar, "H.264 Macroblock Line Level Parallel Video Decoding on Embedded Multicore Processors," Digital System Design (DSD), 2012 15th Euromicro Conference on 5-8 Sept. 2012

[20] J-Y. Chen, C-W. Chiu, G-L. Li, M-J. Chen "Burst-Aware Dynamic Rate Control for H.264/AVC Video Streaming," Broadcasting, IEEE Transactions on March 2011

[21] X. Chun-Dong, L. Yi-Qing, Z. Mu-Gui, "A Design of Embedded Video Server Based on H.264," Communication Systems and Network Technologies (CSNT), 2012 International Conference on 11-13 May 2012

[22] E.S.C.Takahashi, Application aware scheduling for power management on IEEE 802.11, Proc. IPCCC., Feb. 2000,

[23] W. Pu, Y. Lu, F. Wu, "Joint Power-Distortion Optimization on Devices with MPEG-4 AVC/H.264 Codec," International Conference on Communications (ICC), vol. 1, pp 441-446, 2006

[24] Z. Chen, P. Zhou, Y. He, "Fast Integer Pel and Fractional Pel Motion Estimation for JVT," Joint Video Team (JVT) of ISO/IEC MPEG and ITU-T VCEG, Awaji, Island, JP, 5-13 December, 2002

[25] Conti, J.P. , "Talk about a change [VoIP replacing PSTN," IET JOURNALS and MAGAZINES, Volume: 50, Issue: 12, 2004

[26] Z. Stojanovic, D. Babic, "Traffic Engineering for VoIP Network based on PSTN Statistical Models," 9th International Conferences on Telecommunication in Modern Satellite, Cable, and Broadcasting Services (TELSIK09), 564-567, October 2009

[27] Y. Zhang, SIP-based VoIP network and its interworking with the PSTN, Electron. Commun. Eng. J., vol. 14, no. 6, pp. 273282, Dec. 2002.

[28] B. Ngamwongwattana, A. Sombun "Enhancing Adaptive Control in VoIP by Characterizing Congestive Packet Loss," 11th International Conference on Advanced Communication Technology, 2009.

[29] L.Caviglione, "A Simple Neural Framework for Bandwidth Reservation of VoIP Communications in Cost-Effective Devices ," IEEE Transactions on Consumer Electronics, Vol. 56, No. 3, August 2010

[30] P. C. K. Hung, M. V. Martin "Scurity issues in VoIP applications," Canadian Conference on Electrical and Computer Engineering, 2006. CCECE '06.

[31] D. Butcher, X. Li, J. Guo "Security Challenge and Defense in VoIP Infrastructures," IEEE TRANSACTIONS ON SYSTEMS, MAN, AND CYBERNETICSPART C: APPLICATIONS AND REVIEWS, VOL. 37, NO. 6, NOVEMBER 2007

[32] Salami, R. and Laflamme, C. and Adoul, J.-P. and Kataoka, A. and Hayashi, S. and Moriya, T. and Lamblin, C. and Massaloux, D. and Proust, S. and Kroon, P. and Shoham, Y. "Design and description of CS-ACELP: a toll quality 8 kb/s speech coder," Speech and Audio Processing, IEEE Transactions on 1998

[33] A. Clark, "Modeling the Effects of Burst Packet Loss and Recency on Subjective Voice Quality," 2nd IPTel Workshop, 2001, pp.123 127.

[34] www.ti.com, Texas Instruments company

[35] Linley Gwennap, "BRCM 5000 Provides Dual Issue, Multithreading for Broadband Chips," The Linley group, www.MPRonline.com, 2010