Quality of Service for VoIP in Wireless Communications

by

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Abstract

EVER since telephone services were available to the public, technologies have evolved to more efficient methods of handling phone calls. Originally circuit switched networks were a breakthrough for voice services, but today most technologies have adopted packet switched networks, improving efficiency at a cost of Quality of Service (QoS). A good example of packet switched network is the Internet, a resource created to handle data over an Internet Protocol (IP) that can handle voice services, known as the Voice over the Internet Protocol (VoIP).

The combination of wireless networks and free VoIP services is very popular, however its limitations in security and network overload are still a handicap for most practical applications. This thesis investigates network performance under VoIP sessions. The aim is to compare the performance of a variety of audio codecs that diminishes the impact of VoIP in the network. Therefore the contribution of this research is twofold: To study and analyse the extension of speech quality predictors by a new speech quality model to accurately estimate whether the network can handle a VoIP session or not and to implement a new application of network coding for VoIP to increase throughput.

The analysis and study of speech quality predictors is based on the mathematical model developed by the E-model. A case study of an embedded Session Initiation Protocol (SIP) proxy, merged with a Media Gateway that bridges mobile networks to wired networks has been developed to understand its effects on QoS. Experimental speech quality measurements under wired and wireless scenarios were compared with the mathematical speech predictor resulting in an extended mathematical solution of the E-model. A new speech quality model for cascaded networks was designed and implemented out of this research. Provided that each channel is modelled by a Markov Chain packet loss model the methodology can predict expected speech quality and inform the QoS manager to take action.

From a data rate perspective a VoIP session has a very specific characteristic; exchanged data between two end nodes is often symmetrical. This opens up a new opportunity for centralised VoIP sessions where network coding techniques can be applied to increase throughput performance at the channel. An application layer has been implemented based on network coding, fully compatible with existing protocols and successfully achieves the network capacity.

Acknowledgements

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Finally, I dedicate this thesis to my parents Guillermo Lopetegi and Hirune Zinkunegi, to my brothers Guillermo and Lander, and friends with special mention to Lee Morgan-Thomas.

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A/D	-	Analogue to Digital converter
ACK	-	Acknowledgement
AIFS	-	Arbitration inter-frame space
AP	-	Access Point
AR	-	Auto Regression
ATM	-	Asynchronous Transfer Mode
AuC	-	Authentication Centre
AVP	-	Audio Video Protocol
BER	-	Bit Error Rates
BSC	-	Base Station Controller
BSS	-	Basic Service Set
BW	-	Bandwidth

CBR	-	Constant Bit Rate
CELP	-	Code Excited Linear Prediction
CFP	-	Contention Free Period
CM	-	Conventional Method
CP	-	Contention Period
CSMA/CA	-	Carrier Sense Multiple Access protocol with Collision Avoidance
CSN	-	Circuit Switched Network
CSRC	-	Contributing Source Identifier
CTS	-	Clear to Send
CZ	-	Concatenated Zigzag
DCF	-	Distributed Coordination Function
DiffServ	-	Differentiated Services
DIFS	-	DCF inter-frame space
DLL	-	Delay Lower Limit
EIFS	-	Extended inter-framed space
FCS	-	Frame Check Sequence
FEC	-	Forward Error Correction
FR	-	Frame Relay
FSMC	-	Finite State Markov Chain
GSM	-	Global System of Mobile Communications
GW	-	Gateway
HDR	-	Packet Header

HLR	-	Home Location Register
HTTP	-	Hipertext Transfer Protocol
IDE	-	Integrated Design Environment
IEEE	-	Institute of Electrical and Electronic Engineers
IETF	-	Internet Engineering Task Force
IFS	-	Inter-frame space
IntServ	-	Integrated Services
ISP	-	Internet Service Provider
ITU	-	International Union of Telecommunications
LBP	-	Leader Base Protocol
LP	-	Linear Prediction
LPC	-	Linear Prediction Coefficients
LS	-	Location Server
LSTR	-	Listener Sidetone Rating
MC	-	Markov Chains
MCC	-	Mobile Control Centre
MCU	-	Micro-controller unit
MD	-	Minimum Delay
MEGACO	-	Media Gateway Controller
MGCP	-	Media Gateway Control Protocol
MIME	-	Multi-purpose Internet Mail Extensions
MOS	-	Mean Opinion Score

MPLS	-	Multi protocol Label Switching
MS	-	Mobile Stations
MSB	-	Most Significant Bit
MSC	-	Mobile Switching Centre
MSDU	-	MAC Service Data Unit
NAV	-	Network Allocation Vector
NC	-	Network Coding method
NCD	-	Network Coding Decision
OS	-	Operation System
OSI	-	Open System Interconnection
PAMS	-	Perceptual Analysis Measurement System
PBX	-	Private Branch Extensions
PCF	-	Point Coordination Function
PCM	-	Pulse Code Modulation
PDF	-	Probability Distribution Function
PER	-	Packet Error Rates
PESQ	-	Perceptual Evaluation Speech Quality
PIFS	-	PCF inter-frame interval
PL	-	Payload
PLCP	-	Physical Layer Convergence Protocol
POTS	-	Plain Old Telephony System
Ppc	-	Packets Per Conversation

PPDU	-	PLCP Data Unit
PSN	-	Packet Switched Network
PSTN	-	Public Switched Telephone
QDU	-	Quantization Distortion Units
QoS	-	Quality of Service
RFC	-	Request for Comments
RLR	-	Receiver Loudness Rating
RPE-LTP	-	Regular-pulse-excited long-term prediction
RS	-	Redirect Server
RSVP	-	Resource Reservation Protocol
RTCP	-	Real Time Control Protocol
RTP	-	Real time Transport Protocol
RTS	-	Request to Sent
RTT	-	Round Trip Time
SDP	-	Session Description Protocol
SFD	-	Start of Frame Delimiter
SIFS	-	Short Inter-frame Space
SIP	-	Session Initiation Protocol
SLR	-	Sender Loudness Rating
SOHO	-	Small Office/Home Office
SS7	-	Signalling System 7
SSRC	-	Synchronisation source identifiers

STMR - Side tone Masking Rating

TELR - Talker Echo Loudness Rating

UA - User Agent

UAC - User Agent Client

UAS - User Agent Server

UDP - User Datagram Protocol

URI - Uniform Resource Identifier

VLR - Visitor Location Register

VoIP - Voice over Internet Protocol

WAN - Wide Area Network

List of Symbols -

γ - Information rate received by each sink

T_{xor} - Time to compute the operation itself which is dependant on hardware processors

ρ - Channel factor for concatenated systems with different audio codecs

η - FEC efficiency

$\mathbb{F}$ - Field

τ - Refer to the number of independent packets required to perform network coding

$\vec{x}$ - Source information vector

A - Advantage factor

Bpl - Packet-loss robustness

$BurstR$	-	Average length of observed burst in an arrival sequence over the average length of a expected burst under a random loss channel
C_{calls}	-	Capacity of the network to handle VoIP calls within a BSS
C_{ij}	-	Capacity of an edge
CW_{min}	-	Minimum contention window
Ds	-	Sender distortion value of the telephon
E	-	Edges
F	-	The flow of the graph
f_{ij}	-	The information units that are sent from node i to node j
G	-	Graph
I_D	-	Delay impairment factor
I_e	-	Equipment impairment factor
I_S	-	Simultaneous impairment factor
I_{dd}	-	Impairment for too long absolute delay
I_{dle}	-	Listener Echo impairment
I_{dte}	-	Talker Echo impairment
I_{e-eff}	-	Effective equipment impairment factor
I_{eq}	-	Distortion equaliser impairment factor
$M_{i,j}$	-	Encoder matrix for node i connected to j number of output nodes
N_0	-	Sum of all noises
Nc	-	Circuit noise
P	-	Probability of being in one state
p	-	Probability of moving from a GOOD state to BAD state

Pn - Packets sent from a single VoIP UAC

Ppl - Packet-loss probability

Pr - Room noise at the receiver side

Ps - Room noise at the sender side

q - Probability of moving from a BAD state to GOOD state

q_{min} - The minimum number of packets that has to be in the queue to perform the modulo two addition

R - R factor

R_0 - Signal to noise impairment factor

R_{AP} - Data rate agreed between a the AP and the mobile stations

s - Source node

T - One way delay

t - Sink node

T_a - absolute delay

T_g - Packetization delay

T_p - Conventional single VoIP packet delay

T_r - Round trip delay

T_s - Refers to slot time

T_t - Total transmission delay of one voice packet per user

t_w - Waiting time for efficient network coding design for VoIP communications

T_{arr} - Delay introduced by the waiting time to receive a second packet to perform the modulo two addition of the packets

$T_{backoff}$	-	Back-off time period introduced by IEEE 802.11b to prevent collisions
T_{data}	-	Delay introduced by the IP/UDP/RTP/Payload
T_{ldata}	-	Time associated with larger packet data due to synchronisation
T_{max}	-	Maximum waiting time allowed by the algorithm to perform network coding at at the AP
T_{ntw}	-	Delay introduced by network coding,
T_{oh}	-	Delay introduced by the MAC overhead of the packet
T_{sync}	-	Time associated with additional header introduced in network coding to achieve synchronisation
V	-	Vertices
X_d	-	Input Symbols to a node
z_i	-	Symbol over a field $\mathbb{F}$

CHAPTER 1

Introduction

1.1 Introduction

VOICE over Internet Protocol (VoIP) substantially increases traffic load and can become a problem if it is not addressed in advance. One possible but expensive solution would be to set up an entire new data network dedicated to VoIP. A more sensible approach however is to combine the data network with the VoIP service which requires a monitoring system if customer satisfaction has to be delivered.

Quality of Service (QoS) is the requirement to guarantee good customer experience as well as fairness in network allocation, but what is a good customer experience? The International Union of Telecommunications (ITU) has developed a recommendation called the Mean Opinion Score (MOS) [1] describing a procedure to validate voice quality perception throughout large number of customers assessment. A more efficient approach is to use known communication parameters to set a mathematical model to predict VoIP call quality. For example, if parameters such as delay and packet losses are known, a prediction of voice quality can be computed using the E-model [2]. This model analytically clarifies the feedback received from the MOS experiments, consequently becoming apparent that the prediction of phone quality is feasible and very important aspect when designing more efficient network architectures. This is the main reason why in the last decade QoS has become such important topic since it is known that the customer is always right.

Managing VoIP in wire networks with QoS is already a challenge [3], thus the emergence of wireless networks, whilst making it more attractive to customers, has increased complexity in allocating resources to guarantee good quality. At the same time, customer awareness of the technological issues related to wireless networks has brought down the expectations of a good quality VoIP experience. This fact opens an opportunity to network managers to use more efficient though less accurate audio codecs with higher compression rates to maximise the throughput of the network. Hence, in this thesis, speech prediction is considered when improving VoIP communications implementing a test-bench for wire to wireless networks with Forward Error Correction (FEC) algorithms. Furthermore, the current E-model is extended to predict VoIP quality over heterogeneous networks which is a key feature for future network design.

Whilst researching these future architecture designs, network coding will play an important part as its major achievement, the gain in network throughput, has shown great potential to change the paradigm of broadcast communications. At the moment, conventional nodes within the network are initially set to route information according to the shortest path or fixed tables since this method focuses on maximising the highest throughput of point to point communication. A new point of view, is to consider network resources as a whole rather than a single path to the destination. Hence, the new system has multiple information sources to be delivered to multiple destinations thus creating opportunities to insert encoding methods at nodes that were previously just *storing and forwarding* information. In this thesis, network coding schemes for VoIP sessions over a Basic Service Set have been designed and implemented in a hardware platform to investigate the performance of these methods over real-time services.

1.2 Motivation

Most research in VoIP has been dedicated to the study of performance from a single network point of view but real world network and usage scenarios are more likely to be deployed as depicted in Figure 1.1. This illustration shows a global network where customers are connected via different technologies; *Example A* represents a VoIP call from a soft-phone to a wireless connected cell phone, *example B* reflects the communication between two wireless nodes and *example C* is a soft-phone to soft-phone connection through a Wide Area Network (WAN). The cross interaction shown in these examples has an impact on call quality performance. For example, consider *example A* where a soft-phone might be using a low data rate audio codec and the receiver in a wireless network has a fixed audio codec. In this case, the Gateway (GW) has to decode the codec only to encode again according to the specifications of that network and in doing so becomes a critical node of the system. If *example B* is considered, mobility issues related to channel performance or handover can lead to an increase on delay and packet loss rates becoming the base station a bottleneck. Finally the WAN connection in *example C* can be exposed to packet drops whenever Internet servers have to deal with large queues. Hence, from a network designer perspective, it is crucial to know how a network will perform before allocating the required resources. An experimental case scenario and a deep understanding of speech prediction models under heterogeneous networks can give an answer to this problem. Furthermore, in order for future networks to become more efficient, intermediate nodes will have to

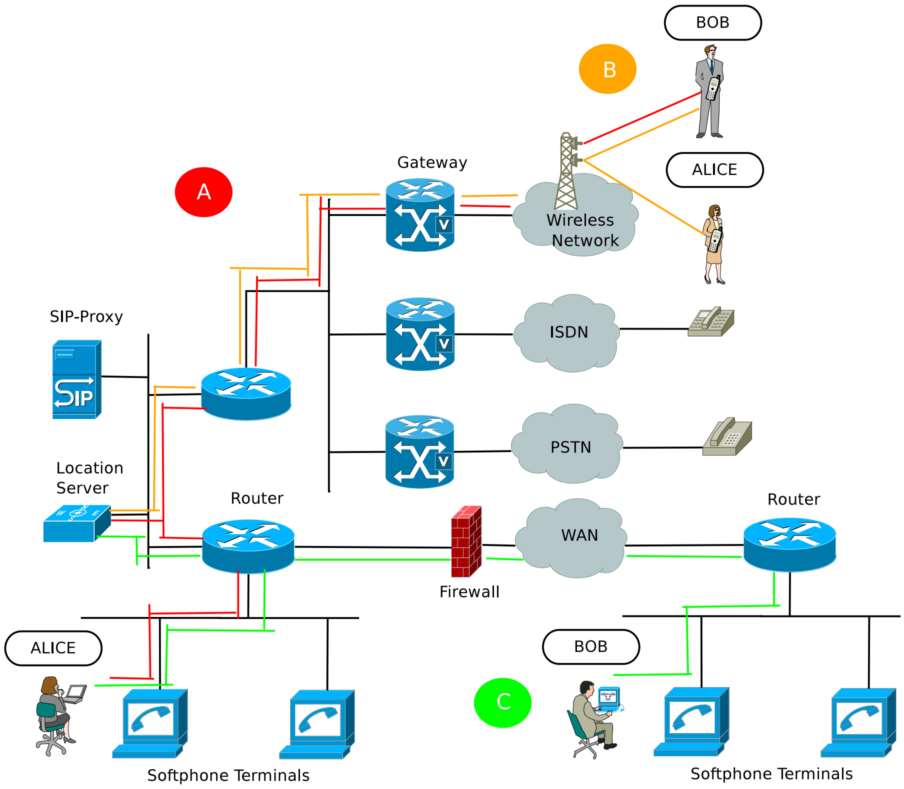


Figure 1.1: Example of a current network topology integrating a variety of technologies.

take a more active role than simply *store, translate and forward* the information.

1.3 Aims and Objectives

The aim of this project is to investigate the prediction of VoIP performance over heterogeneous networks incorporating forward error correction and network coding schemes to real-case scenarios. A SIP proxy GW to a mobile network is implemented as starting point of this research where MOS tests are performed to assess VoIP calls. This implementation is improved by developing active VoIP nodes with awareness of end-to-end QoS. The results obtained are used to design and implement a new mathematical model based on the current E-model [2] to predict speech performance under heterogeneous networks. This thesis is also focused on improving VoIP efficiency. As a matter of fact, research is performed on network coding schemes to be applied to VoIP systems. The investigation aims to improve substantially throughput and delay

performance of VoIP calls over wireless broadcast networks without dependency of the physical layer.
The primary objectives of this research can be summarised as follows:

- Understand and investigate on SIP proxy GWs to connect wire networks to wireless networks with special attention to Small Office Home Office (SOHO) applications.
- Implement an algorithm that improves speech quality performance over bursty networks using active SIP proxy GWs in a test-bench platform.
- Investigate analytical models to assess VoIP quality performance.
- Expand current analytical models to predict VoIP quality over a number of heterogeneous networks.
- Understand and investigate network coding schemes
- Design and implement in hardware, a network coding based VoIP system that improves the throughput performance without undermining QoS criteria.

1.4 Statement of Originality

This research has designed and created an embedded system test-bed for a portable SIP Proxy Gateway to diagnose QoS requirements. This system adopted a FEC algorithm, therefore becoming an active node capable of outperforming conventional GW but at the cost of extra throughput. The study of speech quality from this research resulted in an analytical model to extend the E-model [2] predicting VoIP performance under concatenated networks. Within this procedure, new codecs based on Linear Prediction (LP) have been tested and parametrised to fit the E-model standard. The proposed model adds a new equalization impairment factor to distinguish between poor and bad speech prediction and in addition a methodology to achieve good prediction under two channel communication with a different audio codec. The importance of forecasting VoIP performance under different case scenarios brings the last contribution of this thesis: VoIP with network coding. Recent research carried out in the field of *Information Theory* suggests that network coding can increase the throughput of a network. In this last chapter, an application layer based VoIP system has been designed and developed to prove the efficiency of network coding. The proposed system concludes that the same speech quality can be obtained even with a larger number of users in a Basic Service Set.

1.5 Publications arising from this research

All technical chapters presented in this thesis resulted in a number of tutorials related to VoIP implementation as well as the following IEEE papers:

1. I. Lopetegui , R. A. Carrasco and S. Boussakta, "Speech Quality Prediction in VoIP Concatenating Multiple Markov-Based Channels", in *Proc. IEEE 6th Advanced International Conference on Telecommunications (AICT 10)*, May 2010, pp 226-230.
2. I. Lopetegui , R. A. Carrasco and S. Boussakta, "Embedded Implementation of a SIP Server Gateway with Forward Error Correction to a Mobile Network", in *Proc. IEEE 10th International Conference on Computer and Information Technology (CIT 10)*, June 2010, pp 2415 -2420.
3. I. Lopetegui , R. A. Carrasco and S. Boussakta, "VoIP design and implementation with network coding schemes for wireless networks", in *Proc. IEEE 7th International Symposium on Communication Systems, Networks and Digital Signal Processing (CSNDSP10)*,July 2010, pp 857 -861.
4. I. Lopetegui , R. A. Carrasco and S. Boussakta, "Experimental measurements for VoIP with network coding in IEEE 802.11", in *Proc. IEEE 7th International Symposium on Wireless Communication Systems (ISWCS10)*, September 2010, pp795 -799.
5. I. Lopetegui , R. A. Carrasco and S. Boussakta, "Multicasting VoIP packets with network coding", *IEEE Trans. Multimedia*, submitted (February 2011).

1.6 Organization of the Thesis

This thesis is divided into four chapters. Chapter 2 presents a literature review and introduces VoIP architecture including protocols, topologies and references to the audio codecs utilised throughout this thesis. E-model and MOS methods are described to enable understanding the subsequent chapters as part of the QoS measurement methodologies. This chapter is concluded with an introduction to network coding.

In Chapter 3, the implementation of a SIP proxy GW for embedded systems introduces the test bed utilised to perform MOS experiments. The SIP proxy is equipped with FEC correction to overcome a self developed on board packet loss model based on Markov Chain (MC) theory. Results are compared and discussed with a conventional SIP proxy GW resulting in a favourable performance for the active SIP proxy GW. This study leads to a speech prediction model, explained in the following chapter.

Chapter 4 proposes an extension to the E-model to predict speech quality under heterogeneous networks. The audio codecs chosen at Chapter 2 are tested to measure their equipment impairment factor and robustness. These audio codecs are applied into a set of experiments including two independent networks where all possible combinations are carried out and assessed by the MOS. The experimental results are utilised to investigate an analytical formula in order to achieve the lowest error-margin possible between the subjective speech quality predictor and the mathematical model. The mathematical model and its results are discussed showing a high accuracy with very low error factor.

In Chapter 5, the understanding of speech prediction is used to assess the impact of a network coding based VoIP system. A new VoIP design based on an application layer network coding is described. The design is implemented to corroborate the advantages of network coding in a single Basic Service Set (BSS) where the results reveal a great improvement in throughput performance. These results are discussed alongside speech prediction algorithms where a larger number of VoIP calls have been performed without loosing speech quality performance.

Finally, the conclusion of the research along with the contributions from this thesis is presented in Chapter 6, followed by a set of possible future research directions.

CHAPTER 2

Literature Review

2.1 Introduction

THIS chapter provides a general overview of VoIP protocols, quality of service algorithms and network coding schemes. In Section 2.2, Session Initiation Protocol (SIP) and Real time Transport Protocol (RTP) are described and it is followed by an insight of utilised audio codecs in this thesis. In Section 2.3, both subjective and objective methods to assess speech quality are discussed. Section 2.4 gives an extensive overview of network coding fundamentals and finally Section 2.5 concludes the chapter.

2.2 VoIP Protocols

Voice over Internet Protocol (VoIP) is a technology that can be divided in two fundamental theoretical aspects: the signalling system and the delivery of audio packets. The former refers to the signalling protocol responsible to initiate, control and finish a session of a voice call. The latter is the protocol for transferring voice data through the network. There are a number of alternatives developed for each cornerstone but overall a synergy of protocols around SIP and RTP have become the most popular ones due to its simplicity.

2.2.1 Signalling Protocol: SIP

Originally the signalling systems for VoIP was taken from the Signalling System 7 (SS7) and adapted to the packet switched domain. The development started with the Media Gateway Control Protocol (MGCP), informally defined in Request for Comments (RFC) 3435 [4] that later evolved into Media Gateway Controller (MEGACO) [5]. These protocols are based on a master/slave architecture oriented to the integration of the Public Switched Telephone (PSTN) and VoIP. Further developments led to a Peer-to-Peer orientated signalling system with two known protocols: H.323 and SIP. H.323 is designed to work with local and wide area networks that do not guarantee QoS. It was developed by the ITU-T and unifies older standards into a single one [6]. The high complexity and scalability of H.323, made SIP an easier and faster approach as an alternative to the signalling system over Packet Switched Networks (PSN).

SIP [7] has been developed and planned within the Internet Engineering Task Force (IETF). The protocol has been designed with easy implementation, good scalability, and flexibility in mind. SIP is specified mostly in the RFC 3261 and it defines the creation, modification and termination of a session with one or more participants. A session is a set of senders and receivers that communicate the state kept in those senders and receivers during the communication. These entities with SIP support are defined as User Agents (UA). Each UA is self-sufficient in creating a session with any other node of the network. Hence, it is said that each node has two key components, User Agent Server (UAS) and User Agent Client (UAC). The UAS handles any connection request whereas UAC is responsible for creating any new connection. Four type of entities are defined to route different UA within the network [7]:

- *Location Server* (LS): A service used by a SIP Proxy server to obtain information regarding callees possible location(s). Sometimes can be found within the SIP Proxy server.
- *Proxy Server* : An intermediary program that acts as both a server and a client for the purpose of making requests on behalf of other clients.
- *Redirect Server* (RS): A server that accepts a SIP request, maps the address into zero or more new addresses, and returns these addresses to the client.
- *Gateway Server* (GW): A intermediate server that acts on behalf of the SIP agent to facilitate access to other existing technologies that do not support SIP signalling system.

SIP is designed to interact with nodes in the same way as Hipertext Transfer Protocol (HTTP) does, i.e. in a request-response method [8]. Hence, SIP defines six request methods,

- REGISTER: used to register the users or a third party to the servers.
- INVITE: initiates the call signalling sequence.
- ACK and CANCEL: used to support session setup.
- BYE: terminates a session.
- OPTIONS: queries a server about its capabilities.

Request methods are replied with one of the following six main response codes,

- 1xx: Provisional responses, for instance *180 Ringing*.
- 2xx: Dialogue acceptance response, for instance *200 OK*.
- 3xx: Response with the new address where the UA might be reached, for instance *302 Moved*.
- 4xx: Non accepted final response for a dialogue with information for resubmission the request, for instance *407 Proxy Authentication Required*.
- 5xx: Non accepted final response for a dialogue due to server failure, for instance *503 Service Unavailable*.
- 6xx: Non accepted final response for a dialogue although the server has all the information to proceed with the request, for instance *603 Decline*.

SIP uses Uniform Resource Identifier (URI) [9] to identify a logical destination instead of using IP addresses. It consists of three parts; the protocol that communicates UAs with the SIP server, the name of the server (domain.com) and the name of the resource. The name of the resource can be defined as an email address, telephone number or nickname. An example of that could be *SIP:Bob@sip.ncl.ac.uk*. Additionally, a SIP message includes other headers such as *To*, *From*, *Via* etc. to facilitate the description of the request. The readers are referred to [7] for further details on the subject.

Digital era facilitates audio compressions of different data rate to maximise the throughput performance of VoIP sessions. This implies that if two UAs are about to start a VoIP session both have to support the same audio codec since asymmetrical audio codecs are not allowed. Session Description Protocol (SDP) [10] is commonly used along with SIP to inform other UAs supported audio/video codecs. SDP is a challenge/response method based on priority ordering, i.e. the order of the offered audio codec matters. An example of an SDP message is shown below.

v= 0
o= -7 2 IN IP4 10.12.9.10
s= myphone
c= IN IP4 10.12.9.10

m= audio 65312 RTP/AVP 0 3 38
a= rtpmap:38 SPEEX mode/4

There are five main headers in this example, *v,o,s,c,m* standing for version, origin-field, session name, connection-field and media. Version header is a single digit value and origin-field specifies four parameters: user name space, *-7*, session identification , *2*, network type, *IN IP4* and address of sender, *10.12.9.10*. Session name refers to the name of the phone and media header defines supported audio/video codecs. In this case *audio* is supported at port *65312* with RTP protocol running as an Audio Video Protocol (AVP), followed by a set of numbers that describe the audio codecs supported by the node in priority order. The audio codecs ciphering are defined in [11] where *0* and *3* stand for PCMu and GSM codecs. SDP can add other codecs not specified in RFC 1890 by using Multi-purpose Internet Mail Extensions (MIME) [12]. Such example is the codec defined here as *38*. The codec description is followed by an optional attribute *a= rtpmap: 38 SPEEX mode/4*, where rtpmap defines the format and parameters of Speex codec including the supported mode, i.e. *mode/4*. Further details on audio codecs are covered at Section 2.2.3.

2.2.1.1 SIP routing methods

Depending on the request methods there are three SIP models for connecting UAs: Method 1, Method 2 and Method 3 [7].

Method 1: In this method, two UAs take part in the communication process as seen in Figure 2.1 on the following page and no intermediate nodes are used. Note that in this case the UA does not need to be registered in any place. The *Invite* message is a session initialization packet from SIP in which UAC sends the information such as who this invite is sent from, to whom and which SDP is supported for the communication. Once this packet is received by the recipient, a *180 Ringing* message is sent saying that the phone of the destination is ringing. If the recipient answers the phone call, an *200 OK* is sent with the SDP features that the recipient can support. If the caller agrees, an ACK is sent and RTP packets are exchanged. The phone call is finished by sending a *Bye* message.

Method 2: In the following method, the UA is registered in a LS connected to a SIP Proxy as shown in Figure 2.2 on the next page. Since both servers are connected,

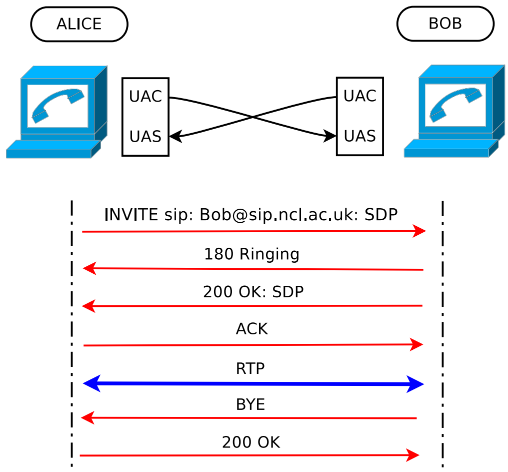


Figure 2.1: SIP method 1.

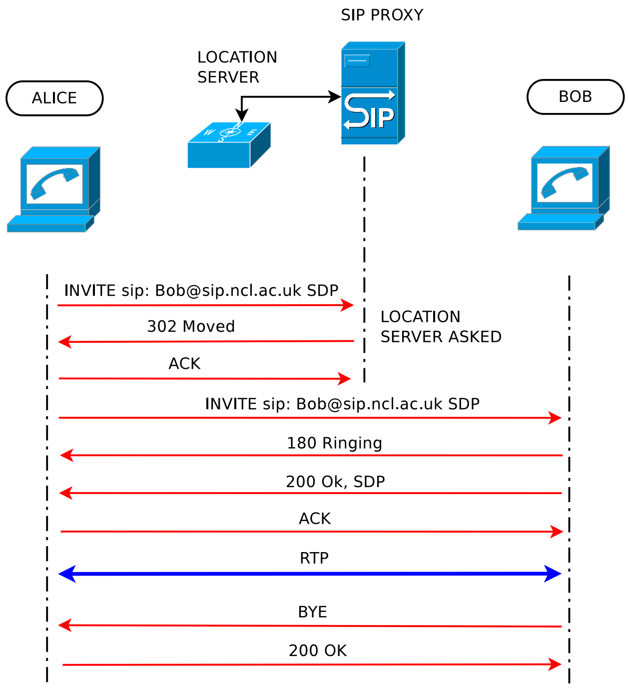


Figure 2.2: SIP method 2.

the caller is redirected to the destination with a *302 Moved* response message. Once the UAC learns the new destination, it reproduces the request with the new information following the same steps as method 1. With this method if a SIP account moves from one network to another it can be tracked down. This is a more scalable choice than method 1 since *Alice* does not need to know where *Bob* is located.

Method 3: In this method, the SIP Proxy interacts with the LS to generate a parallel *Invite* request on behalf of *Alice* to *Bob* as illustrated in Figure 2.3. Thus, the SIP proxy is responsible for finding the next hop, which in this case is the destination itself, and routing all pertinent messages. Once the end user is found, packets are routed via the SIP proxy which monitors the process of the entire conversation. This method is a general case of interaction where *Bob* and *Alice* are part of a heterogeneous network, where the bridging of the networks is carried out with a SIP proxy and a Media Gateway.

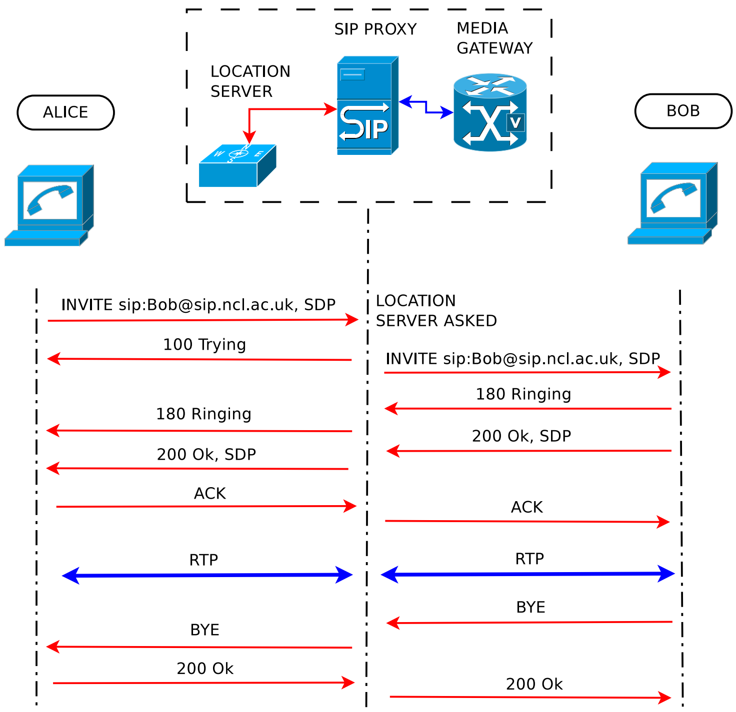


Figure 2.3: SIP method 3.

2.2.1.2 SIP Proxy and Media Gateway

SIP protocol is designed to work with PSN but not all PSN nodes use SIP protocol nor all networks are PSN. If a universal communication system based on PSN is to take over current telecommunication networks, it has to provide gateways to interact with other technologies. The entity responsible of that interaction within SIP is the SIP Proxy with a merged Media Gateway server.

Two main distinction can be made within these servers: SIPs that interact within PSN or within Circuit Switched Networks (CSN). The former is an example of a gateway that interacts between H.323 protocol and SIP. In this case, the signalling system differs from one protocol to another but RTP packets remain the same. Thus as long as both parties agree on the audio codec parameters the call established is essentially a VoIP call over a PSN. The latter requires a more thorough understanding of the CSN technology. Consider that a phone call from the Plain Old Telephony System (POTS) is routed over a PSN. At some point of the network the Gateway has to convert SS7 signalling system's messages into SIP messages. In addition, once the call is established, voice from the POTS has to be packetised and converted to an agreed audio codec between the Gateway and the UA at the PSN side. Consequently the Gateway server becomes a potential bottle neck of the communication system if it does not deliver packets fast enough. In this thesis, Chapter 3 presents an analysis and implementation of a SIP proxy with a merged Media Gateway on an embedded platform to connect IEEE 802.3 and Global System of Mobile Communications (GSM) network which leads to a speech prediction model in Chapter 4. The GSM is a mobile network specifically designed for voice communication and thus it is based on a CSN. Although, fourth generation mobile phone networks are already under development, there are many services that might not be adopted yet to PSN. For example, the European Commission for Rail communications requires the use of GSM networks as a communication system for trains [13]. Equally emergency communication systems such as lift communications require a robust network for critical calls, where GSM is a possible solution fulfilling the requirements of the European Norm EN 81-28 [14]. In conclusion the need of interaction between PSN and CSN is a subject in an ongoing study and it is also analysed in this thesis.

2.2.2 Real-Time Protocol

Voice packets are moved from source to destination with Real-time Transport Protocol (RTP) [15] and controlled by Real Time Control Protocol (RTCP) [16]. RTP provides end-to-end delivery services with a header specification as illustrated in Figure 2.4 on the following page. The header has initially two octets describing session's version, padding, extension, number of Contributing Source Identifier (CSRC), marker and Payload Type. The first four bytes are completed by the sequence number of the packet. The header is followed by a time stamp, Synchronisation Source (SSRC) identifiers and CSRC. This last identifier defines the source identifier of the packet which in this thesis is always set to zero, leaving a 12 byte packet header. RTP typically runs on top of User Datagram Protocol (UDP), although the specification is general enough to support other transport protocols. RTP does not intrinsically provide any mechanism to ensure timely delivery or any QoS. It relies on lower-layers to prevent out-of order packets and delivery acknowledgement. Thus, in an IP network voice commonly travels as IP/UDP/RTP, equivalent to $20 + 8 + 12 = 40$ *bytes* header, without delivery control and maximising the best effort characteristic of the network. Simultaneously, at a default five second frequency, RTCP sends control packets to all participants in the session. Its main function is to offer feedback on the quality of the data distribution, having the chance to advise RTP of any features that may have to be changed.

```
Bytes  0                   1                   2                   3
 Bits   0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
       +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
       |V=2|P|X|  CC   |M|     PT      |       sequence number         |
       +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
       |                           timestamp                           |
       +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
       |           synchronization source (SSRC) identifier            |
       +=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
       |            contributing source (CSRC) identifiers   ....      |
       +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
       |                          AUDIO DATA                           |
       +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 2.4: RTP packet header. V = version, P = Padding, X = extension, CC = CSRC Count , M = Marker, PT = Payload Type.

2.2.3 Audio Codecs

Audio codecs play a very important role in VoIP performance. There are three type of audio codecs: Waveform codecs, parametric codecs and hybrid codecs. Waveform codecs shape the original message by digital values, resulting in a high quality performance for high bit-rate coding. One example of this is Pulse Code Modulation

(PCM) [17]. Parametric codecs estimate speech signal based on digital models and only the parameters of such model are encoded in the bit-stream. This reproduces a very low bit rate output but the perceptual quality can be very low. A commonly used digital model is the Linear Prediction (LP) model that uses a time variant filter whereby the parameters that define the filter are encoded in the bit-stream. Finally, hybrid codecs are a combination of both waveform and parametric codecs. Hybrid codecs use digital models, such as LP, with an error correction system that approaches the model to real speech. An example of this model is Code Excited Linear Prediction (CELP). In this thesis, one waveform codec and two hybrid codecs, named as PCM, Regular-Pulse-Excited Long-Term Prediction (RPE-LTP) and CELP are used [17].

All codecs have to sample and quantify voice before any compression is applied. If voice is considered as a band limited signal of 4Khz, by Nyquist theory the sampling rate has to be 8Khz, which is the common choice for VoIP. Samples are quantified according to the resolution of the Analogue to Digital converter, typically 16 or 8 bit per sample. PCM is based on a 8Khz, 8 bit sampling codec resulting in a 64kbps audio codec. Codified audio speech is packetised and delivered by the IP network where the number of samples per packet is a trade-off between efficiency and delay. Using a small packet size minimizes the delay between the parties, but the bandwidth efficiency becomes poor. In the case of PCM, if just a sample is sent within a single RTP packet, the packet delay would be 0.125 ms but the efficiency is 2.43% calculated as follows [3].

RTP/UDP/IP header is equal to 40 bytes equal to 320 bits
Payload is 8 bit, with a Fs=8000 Hz equal to $T_s = 0.125$ ms
Required BW=328(bit)/(0.125 (ms)) = 2.624Mbps
Packet efficiency= (8/328)·100 = 2.43%

It is clear that using a 2.624 Mbps for a PCM encoding voice system does not provide an efficient trade-off. In [18] it is demonstrated that most of the encoding systems need to have around *20 ms* payload to achieve their best relationship between packetising/delay and utilised bandwidth. Both hybrid codecs utilised in this thesis use 8Khz sampling rate, 16 bit quantifying speech input and 20 ms payload size. Equally both of them are based on LP and *Long term LP* models which are described next.

LP is a model based on predicting the signal $x[n]$ by using past samples. LP uses an Auto Regression (AR) model to predict $x[n]$ so that $y[n] = \sum_{i=1}^{N} a_i x[n-i]$ where $y[n]$ is the prediction of $x[n]$, N is the number of samples considered for the prediction and a_i are the coefficients of the prediction, also named Linear Prediction Coefficients (LPC). The error introduced by this model is $e[n] = x[n] - y[n]$ and the objective is to keep this error as low as possible. If LP attempts to model speech based on a variant filter, *Long Term LP* uses the characteristic of pitch period implicit in speech to produce a *Long Term LP gain*. Here different designs have different performance as explained in the following codec by codec description.

Often speech algorithms are confused by patents and names that belong to different standards. In this research, next three codecs have been chosen: G.711, GSM 06.10 and Speex which use PCM, RPE-LTP and CELP algorithms respectively.

2.2.3.1 G.711 codec

G.711 is defined by the ITU-T Recommendation [19] and uses a non-uniform PCM model to take the advantage of the statistical distribution of voice, where large amplitudes diminishes with an increase in audio magnitude. Two algorithms are defined: μ-law and A-law. Since the distortion comparison of these two algorithms is minimum the use of any of them results in a similar performance. Thus, in this thesis μ-law algorithm is used where the input variable x is captured with 14 bits of uniform quantification, and transformed with a memoryless function $f(x)$ that reduces the distortion error for speech as shown next [17].

$$f(x) = A\frac{ln(1+\mu|x|/A)}{ln(1+\mu)}sgn(x), \ |x| <= A \tag{2.1}$$

where A is the input magnitude's peak and μ is a compression control degree. To decode such output, the inverse function is applied by using [17]

$$f^{-1}(y) = \frac{A}{\mu}\left[exp\left(\frac{ln(1+\mu)\cdot|y|}{A}\right) - 1\right]sgn(y), \ |y| <= A \tag{2.2}$$

Real implementation for G.711 adopts a linear approximation through tables with $\mu = 255$ where only 8 Most Significant bits (MSB) are taken into consideration, resulting in a bit rate of 64 kbps at 8Khz. The algorithm library for this thesis is taken from [20].

2.2.3.2 GSM 06.10 codec

GSM06.10 is the codec chosen by Global System for Mobile (GSM) communications and is defined in [21]. The encoder and decoder generic block diagram is shown in Figure 2.5. Speech input is a 16 bit word sampled at 8 Khz that is analysed by the *LP Analysis* block to calculate the LPC. These coefficients are given to the *Predicition error filter* where the difference between original message and predicted values error are minimised. Since not all prediction errors are required for a good quality reconstruction of the speech, only certain values of these are sent. The filtering of these values is carried out by the *regular-pulse excitation scheme* where the signal is down sampled into different sequences and only the amplitude of such signals is sent. This method avoids to send the position of the predictive errors and reduces considerably the output data rate. Long term LP encoders uses both LPC and error

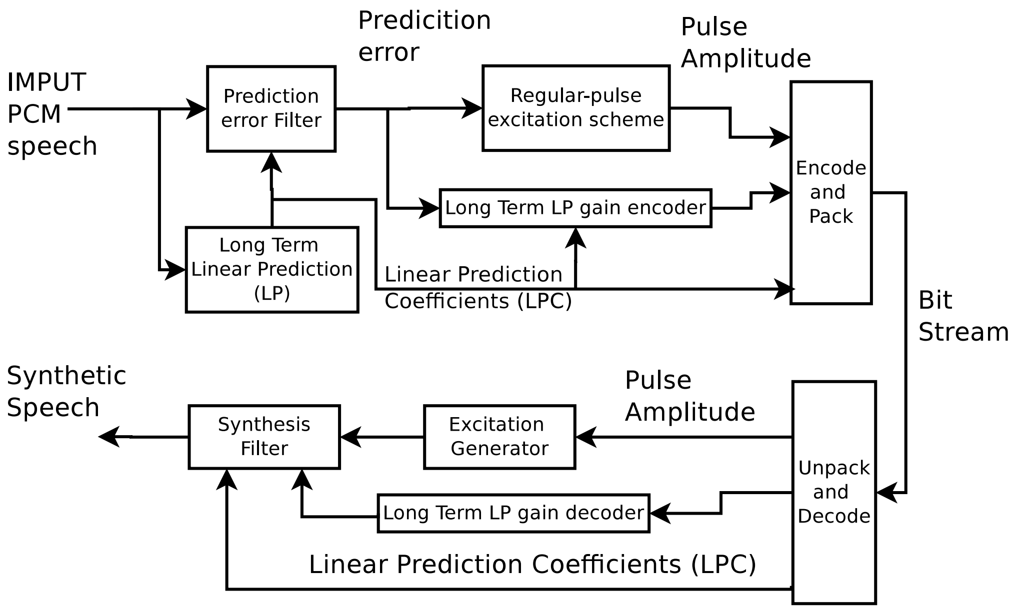


Figure 2.5: GSM 06.10 codec's general block diagram for encoder, *top*, and decoder, *bottom* [17].

prediction to calculate the *Long Term Prediction Gain* index. These outputs are all packed in a 33 byte payload size and transmitted to the decoder. The decoder unpacks the amplitudes of the pulses, the *Long Term Prediction* index and LPC. Pulse amplitudes are fed to the excitation generator to reproduce the encoded predictions.

These predictions are given to the synthesiser that at the same time receives LPCs and the long term gain decoders' output. This output is a set of synthetic speech values in the form of 16 bit PCM samples. The algorithm library for this thesis is taken from [22].

2.2.3.3 Speex codec

Speex is a Code Excited Linear Prediction codec, which implies that the error prediction is minimised with the use of a codebook. Speex codec's block code is presented in Figure 2.6 and full implementation details can be found at [23]. The encoder is composed by an excitation codebook, which can deliver a random noise based on ei-

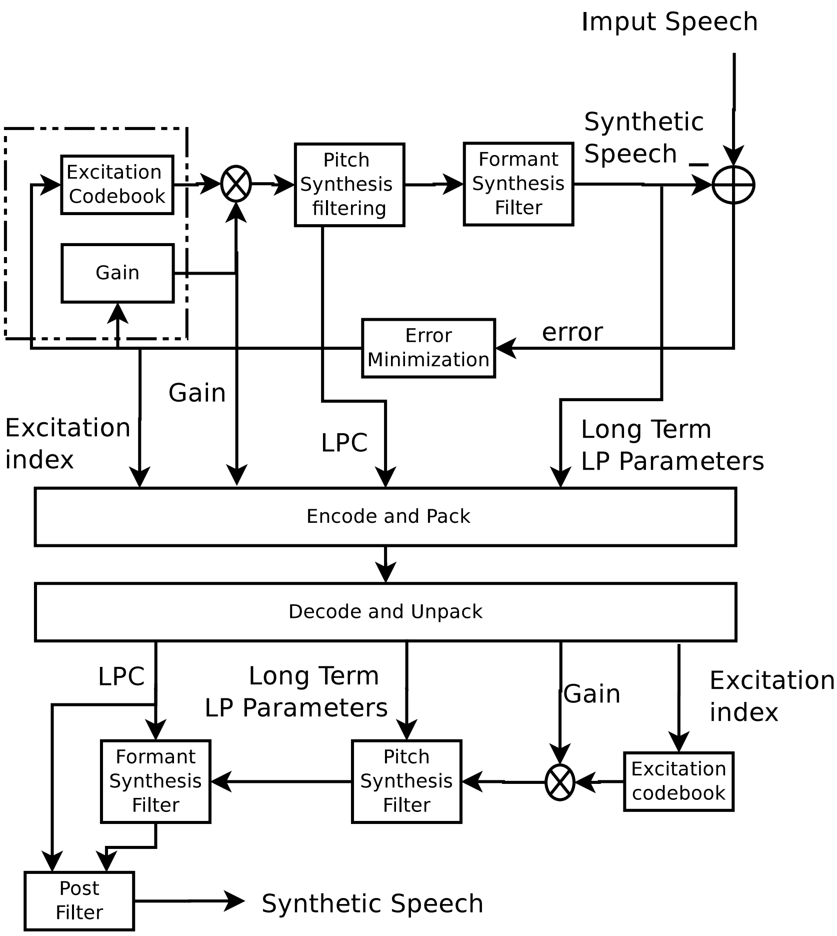


Figure 2.6: Speex codec's general block diagram for encoder, *top*, and decoder, *bottom* [17].

ther fixed or adaptable codebook. This output is then filtered and cascaded with the input gain coefficients relative to the last sample measurement. The pitch synthesis filter is a short term LP that provides the LPCs and creates the periodicity of the signal. The signal is then passed through the formant synthesis filter that gives the envelope to the signal, producing the long term prediction parameters. The output signal is compared with the original speech using a weighted error system where the output is fed back to the gain and codebook. The process is repeated for all codebook's code vectors creating the commonly known analysis by synthesis process of an audio codec. The complexity of the codec and bit rate output is determined by the number of code vectors and therefore it can vary depending on the setup. The process to decode the encoded bit stream follows the inverse steps of the coding process. The difference is that the excitation index facilitates the search of the code vector at the codebook. The gain, LPCs and long term LP parameters are applied to the filters with no major complexity than that of multiplying vectors. The post filter inserted before the synthetic output enhances overall performance by smoothing the increased noise introduced by the error predictor at the encoder side. In this thesis, Speex has been set to *Mode 4* delivering 11,000 bps data rate and the algorithm library is taken from [23].

2.2.3.4 Codec Comparison

The three codecs described above represent a wide range of current audio codecs. G.711 is an example of wideband codec with 64 kbps data rate. Although it is not often used as VoIP audio codec it represents the best voice performance for packet switched networks and therefore is a reference for the rest of codecs. GSM 06.10 and Speex represent narrow band audio codecs. On one side, GSM 06.10 is the officially regulated codec for GSM networks and hence any network willing to interact with it has to support this codec which is the case of the implementation shown in Chapter 3. On the other hand, Speex is part of a last generation of codecs where analysis by synthesis methods are utilised. This codec guarantees good performance over packet losses and high delays, since long term and short term predictions are taken into account and fed back to the error predictor. Current state of the art codecs are often based on CELP.

The performance and characteristics of each audio codec is summarised in Table 2.1 where each encoder's and decoder's delay has been extensively tested, over 20

Table 2.1: Used Codecs Characteristics

Codec	G.711	GSM 06.10	Speex-mod4
Bit rate(kbps)	64	13.2	11.2
Frame interval (ms)	20	20	20
Payload Size (Byte)	160	33	28
Encoding Delay (ms)	0.0231	0.1444	0.4314
Decoding Delay (ms)	0.0169	0.0591	0.0772

times. As expected, codecs with LP have larger encoder values than the waveform based codec.

2.3 VoIP Quality of Service

VoIP technology enables real-time transmission for voice through PSNs using the IP network. This network is known as a Best Effort network because each packet is independently routed to its destination. Hence, packets transmitted by a single source can take different paths to reach the destination while traversing the network. The network performance compromises VoIP in four main aspects: Latency, Queueing and Processing, Packet losses and Jitter [3].

Latency is defined as the delay that the voice data has while crossing the IP network without including any processing or queuing. This parameter is a measure of the propagation delay of the data through the wires.

Queuing and processing is the delay associated with the processing in the router/switches that need to read the IP destination to route the packets to their next hop. Sometimes, as the propagation delay can be assumed as zero, the concept latency is used to express the delay of the propagation plus the queuing and processing delay.

Assuming Latency as the summation of both delays it is necessary to distinguish that in voice communication there are two latencies, one per voice direction. Round trip latency is the summation of the two way latency. The ITU Recommendation G.114 [24] says that the round trip latency range is acceptable when it is less than 150 ms for most user applications. Relatively between 150 and 400 ms is acceptable for administrators that are aware of which connectivity they are using; over 400 ms is unacceptable. Likewise, different latency ranges can cause echo and talker overlap. If the round trip latency is more than 50 ms, the echo from one of the speaker appears in the communication, thus an echo canceller must be implemented in the vocoder. The talker overlap is considered when the one-way delay is greater than 250 ms.

Packet losses occur mainly for two reasons; the first cause is related to a physical layer problem where too many bits are corrupted forcing the receiver to reject the message. The second reason is due to finite memory space at the routers that eventually can not allocate more space for incoming packets failing to deliver the information to its destination.

Jitter is the variation of packets' delay that reach their destination. The variation of inter-packets arrival rate makes the conversation unbearable. This problem is commonly solved by introducing a buffering system to reduce the effect of this feature.

The necessity to guarantee certain QoS might not affect to a well organised network scheme within a LAN. The high speed connection of the typical Ethernet interface of the PC makes the voice communication just as effective as it is with analogue phones. However, to guarantee the same level of QoS over the Internet requires allocating resources that might not belong to the end users any more, inducing the need of extra protocols to overcome the problem. There are two different approaches at this stage.

First option is to offer QoS with protocols working within the IP layer where two alternatives are available: Integrated Services (IntServ) [25] and Differentiated Services (DiffServ) [26]. IntServ is a model that guarantees the QoS between end nodes. To cope with this task every single hop of the network must agree with the requirement of the session initiation host. If these parameters are not satisfied the communication would not be started. IntServ was originally developed by Cisco System and its corresponding signalling protocol is Resource Reservation Protocol (RSVP) [27]. The protocol defines a signalling system to reserve resources for both multicast and unicast communications with a request-response methodology. In DiffServ model, QoS is defined by classifying the traffic and adopting different priorities accordingly. Therefore it is more scalable than IntServ because each router decides the priority associated to the data. However in real case scenarios using DiffServ might not be as effective as it seems. If a company agrees to use DiffServ with an ISP it can not be guaranteed that all data is going to be routed by this ISP's routers. Thus, crossing different ISPs can not guarantee that the priorities are accomplished from the source to the destination, compromising the desirable QoS.

The second option to provide QoS is within the second layer of the OSI model. Three possible options are available; Frame Relay (FR) [28], Asynchronous Transfer Mode (ATM) [29] and Multi protocol Label Switching (MPLS) [30]. FR and MPLS technologies are based on creating virtual circuits on the network and guarantee a minimum bandwidth. ATM differs from the other solutions because it offers different way of organizing the traffic of your network depending on the requirements. If a parallelism is to be made, IntServ is equivalent to FR and MPLS at a higher layer and DiffServ is an ATM/MPLS service above the IP layers. Although these solutions are accessible, they are often expensive or simply not feasible. Subsequently VoIP traffic is often routed on a Best Effort basis without considering network resource allocation. Therefore the constraints of PSNs must be considered to understand the quality performance of VoIP communications as it is shown in the following subsection with subjective and objective methods to assess VoIP calls.

2.3.1 Mean Opinion Score

Mean Opinion Score (MOS) [1] is a subjective methodology in which a large number of people (over twenty according to the recommendation) are interviewed and asked to assess the quality of a voice call. Calls are rated from 5 to 1 where 5 is excellent quality and 1 represents bad quality.

5. EXCELLENT: The call has excellent sound quality resulting in no technical difficulties.
4. GOOD: The call has good sound quality with audio similar to a long distance phone call.
3. FAIR: The call has fair sound quality with some interruptions requiring one or both parties to repeat what they said in the call.
2. POOR: The call has poor sound quality with each party having difficulties hearing the other speak clearly.
1. BAD: The call has such bad sound quality that neither party can communicate effectively.

For instance, Public Switched Networks is a CSN using PCM and scores 4.3 out of

5. The advantage of the method is that obtained results represent real voice quality perception by human beings. Conversely the method is time-consuming and thus alternative methods based on objective measurements are accepted to asses VoIP phone calls.

2.3.2 Speech Quality Predictor: E-model

There are two types of objective methods: intrusive and non-intrusive. On one hand, intrusive methods utilise a reference source to compare it with the degraded signal where the larger the difference the worst is the quality. For example Perceptual Evaluation Speech Quality (PESQ) and Perceptual Analysis Measurement System (PAMS) are based in such method. The problem of this procedure is that the source information or original voice, has to be compared with the received voice which in most cases is impossible since users are physically apart. On the other hand, non-intrusive methods observe parameters that interact with the VoIP call to predict the quality. ITU-T's G.107 defines the E-model [2] where speech quality is assessed by a set of none time-varying additive impairments. This method allows to measure the quality independently where the users are located which is essential for VoIP calls. In this thesis, the E-model has been chosen as VoIP call quality assessment method along with the MOS tests.

The E-model is an objective computational model to assess the transmission variations for a voice conversation of 3.1 KHz. The model is based on a *mouth to ear* system that generates a rating factor namely, *R-value*. The model takes into account impairment factors from an end-to-end perspective to conform the *R-value* which has a range starting from 0 to 100. The model was developed in accordance to extensive subject laboratory tests and it can be transformed to the MOS values with next equation [2].

$$MOS = \begin{cases} 1 & \text{for } R = 0 \\ 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6} & \text{for } 0 < R < 100 \\ 4.5 & \text{for } R \geq 100 \end{cases} \tag{2.3}$$

Figure 2.7 shows the conversion from one model to another. Note that the maximum value of R corresponds to a 4.5 value of the MOS score which means that regardless how good the voice has been digitalised it is never as good as the original.

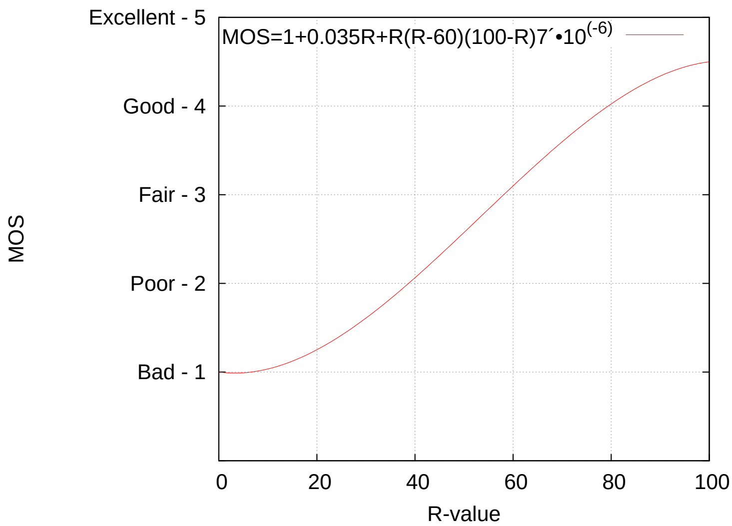


Figure 2.7: Objective method, i.e. *R-value*, versus subjective method, i.e. MOS.

The *R-value* is defined as follows [2]:

$$R = R_0 - I_S - I_D - I_{e-eff} + A \tag{2.4}$$

where R_0 is the basic signal to noise ratio, I_S represents decrease in quality more or less simultaneously with the voice transmission, I_D is responsible for any kind of delay and echo in the system without including the coding delay, I_{e-eff} denotes the impairments due to audio codec and channel constraints and A is the advantage factor related to the service convenience.

Figure 2.8 is the general schematic used by [2] to assess the *R-value*. The signal to noise parameter R_0, describes the noisiness of the systems including: Sender Loudness Rating (SLR), room noise at the sender side (Ps), sender distortion value of the telephone (Ds), Receiver Loudness Rating (RLR), room noise at the receiver side (Pr) and Listener Sidetone Rating (LSTR). Equally, simultaneous impairment factor I_S, depends on signal to noise parameter (R_0), Sender Loudness Rating (SLR), Receiver Loudness Rating (RLR), Side tone Masking Rating (STMR) and Talker Echo Loudness Rating (TELR). The advantage factor, A, refers to the fact that users perception of complex communications affects the expectation of voice quality. For

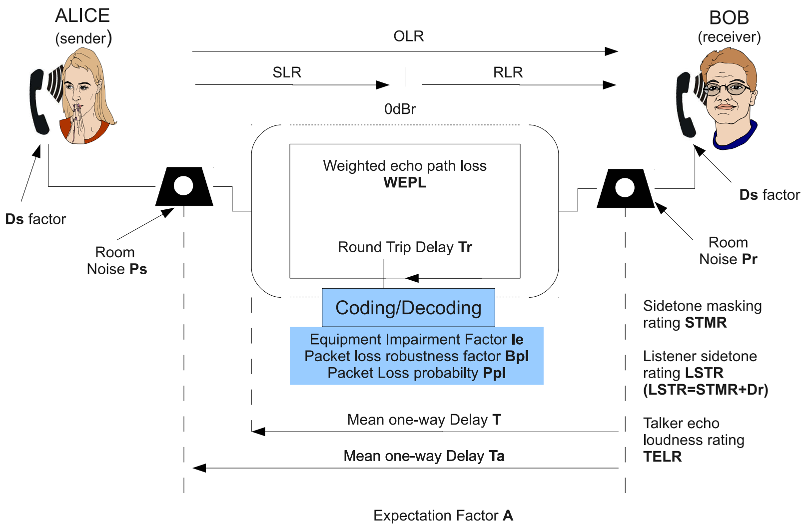


Figure 2.8: E-model [2] communication system.

instance, this factor equalises the *R-value* when users are aware of speaking to somebody in the other side of the Atlantic sea, where greater delays are more tolerated. These parameters described so far, can be difficult to measure and in most of the cases constitute invariant values throughout a call. Thus the ITU-T recommends the use of default parameters (see Appendix A for a detailed description of default values and formulae for these parameters) to facilitate the calculation of VoIP call quality where equation (2.4) can be reduced to [2]

$$R = 93.3 - I_D - I_{e-eff} \tag{2.5}$$

The reduced formula allows us to obtain a valid value of the quality of service of our system focusing on two cornerstones for voice over packetised systems; delay and packet loss. In our case, default parameters are used as specified in Appendix A Table 1 with exception to those parameters referring to delay, channel status and audio codec type. Delay parameters are defined in I_D and are denoted following the figure as follows : T is one-way delay, T_a is the absolute delay and T_r refers to the round trip delay. In this thesis, the approximation of $T_r = 2 \cdot Ta$ is considered valid and is used unless otherwise stated. Channel status and audio codec conform the

I_{e-eff} where Ppl refers to packet-loss probability, Bpl is packet-loss robustness and I_e is the equipment impairment factor. Note that the I_e value is a given parameter by the ITU-T G.113 recommendation [31] whereby codecs are tested under experimental scenarios to assess their performance.

I_D has a different mathematical expression depending on T_a and STMR. While T_a is considered as a variable parameter, STMR is set as a default parameter (15 dB). Setting STMR to a static value means that VoIP prediction is carried out under devices that perform ideally with side tones and echo cancellation. I_D is therefore defined as [2]:

$$I_D = I_{dte} + I_{dle} + I_{dd} \tag{2.6}$$

where I_{dte}, I_{dle} and I_{dd} refer to impairments related to the *Talker Echo*, *Listeners Echo* and too long absolute delay (T_a), respectively. At the same time, I_{dte} is defined as [2]:

For 9dB $\leq$ STMR $\leq$ 20dB

$$I_{dte} = \left(\frac{Roe - Re}{2} + \sqrt[2]{\frac{(Roe - Re)^2}{4} + 100} - 1 \right) \left(1 - e^{-T}\right) \tag{2.7}$$

where, Roe and Re are [2]

$$Roe = -1.5(N_0 - RLR)$$

$$Re = 80 + TERV - 14$$

$$TERV = TELR - 40 \log_{10} \frac{1 + \frac{T}{10}}{1 - \frac{T}{150}} + 6e^{-0.3T^2}$$

$$N_0 = -61.16, \text{ from Appendix A} \tag{2.8}$$

where N_0 refers to the power addition of different noises calculated with R_0 using default values specified in [2] and the rest of parameters are intermediate variables. The I_{dle} factor is calculated using the following equation [2]:

$$I_{dle} = \frac{R_0 - R_{le}}{2} + \sqrt[2]{\frac{(R_0 - R_{le})^2}{4} + 169} \tag{2.9}$$

where $R_0 = 94.74$ is calculated using default values from Appendix A and R_{le} is [2]

$$R_{le} = 10.5 \cdot (WEPL + 7)(T_r + 1)^{-0.25}$$

where WEPL is a default value from Appendix A . Finally I_{dd} is a value that depends upon the absolute delay value, whereby even with perfect echo cancelling the VoIP call can be perturbed. I_{dd} is set to 0 if $T_a \leq 100$ and if not I_{dd} is set to [2]

$$I_{dd} = 25 \left((1 + X^6)^{\frac{1}{6}} - 3 \left(1 + \left(\frac{X}{6} \right)^6 \right)^{\frac{1}{6}} + 2 \right) \tag{2.10}$$

where

$$X = \frac{\log_{10}\left(\frac{T_a}{100}\right)}{\log_{10} 2} \tag{2.11}$$

Following the illustration in Figure 2.8, where it is shown that I_{e-eff} depends on I_e, Ppl and Bpl, the E-model's channel modelling is based on the interpretation of Ppl under different burstiness scenarios. Hence I_{e-eff} is defined as follows [2]:

$$I_{e-eff} = I_e + (95 - I_e) \cdot \frac{Ppl}{\frac{Ppl}{BurstR} + Bpl} \tag{2.12}$$

where BurstR refers to a Burst Ratio, defined as the average length of observed burst in an arrival sequence over the average length of a expected burst under a random loss channel. In other words, if a random channel is considered then $BurstR = 1$ which is a rare case for channel modelling. In the following chapters, channel models based on experimental data are introduced and applied to different experiments to investigate the performance of VoIP under heterogeneous networks.

In order to understand the impact of delay and packet losses, Figure 2.9 shows only delay increasing versus only packet loss rate increasing performance. The prediction is calculated for a G.711 case scenario where $Bpl = 4.3$, $I_e = 0$ and $BurstR = 1$. The illustration shows that the increase of packet loss produces a severe quality drop whereas delay increase has a smoother repercussion. In conclusion, it is visible that a network should consider minimising the packet loss percentage.

The E-model describes a method to estimate the speech quality over different case scenarios, whereby using delay and packet loss parameters, quality performance is assessed. The great advantage of such this method is the rapid evaluation it provides

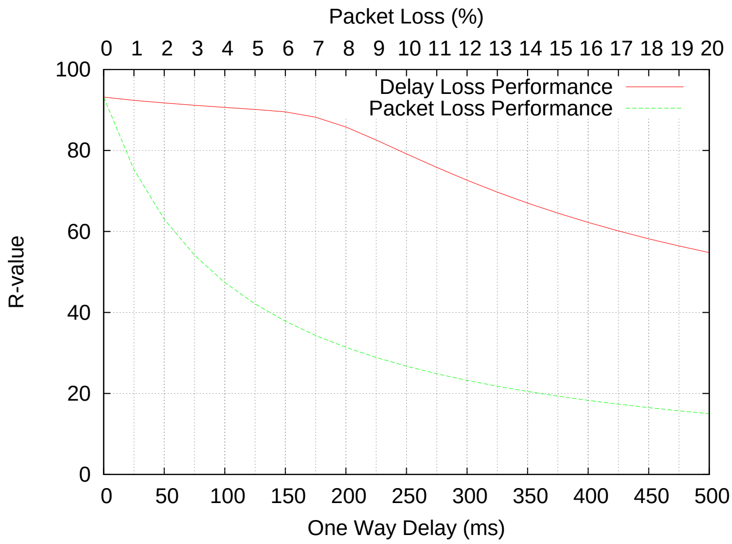


Figure 2.9: *R-value* comparison for a delay only and packet loss only increase scenarios.

to both end users and intermediate nodes. Conversely, the model remains limited. For instance not all codecs are available in [31] nor is the method prepared to handle multiple network channels. For these reasons Chapter 3 and Chapter 4 are dedicated to the implementation and modelling of a SIP servers with two different link layers that cover some of the existing limitations of the model.

2.4 Fundamentals of Network Coding

Previous sections introduced VoIP protocols and QoS assessment methods. In this section, the literature review is concluded with an introduction to a set of schemes that is highly promising to radically change the paradigm of future networks. A recent breakthrough in an Information Theory paper showed that *network coding* techniques can considerably improve throughput efficiency of the channel [32]. Essentially, the theorem states that by combining existing packets in the network, the maximum flow can be achieved, i.e. an encoding system within intermediate nodes can achieve the capacity of the network. The following subsections describe network coding theory and the network topology used in this thesis for a VoIP system with network coding support.

In a point-to-point connection, information theory divides a communication in source and channel coding. The former aims to achieve the highest compression of raw data, good examples are audio codecs used by VoIP services. The latter enables transmission over the channel essentially error free providing that extra information has to be added to the original information. For example, in the Internet itself raw information is packetised (source coding) and transmitted (channel coding) from node to node according to routing tables that provide the next hop address. This *store and forward* method considers source information to be completely independent and if the same information has to be delivered to two different destinations, then the source has to be duplicated. In other words, if there are n receivers, the source information is repeated n times overloading the network considerably. This problem can be solved if intermediate nodes are actively used rather than simply *store and forward* packets where such nodes would combine the incoming information, i.e. *store, code and forward*. The method brings several enhancements: throughput efficiency, robustness to packet loss, reduction of complexity and security [33].

Throughput performance is the best known characteristic of network coding. The butterfly illustration, on Figure 2.10 (a), is an example of a set of nodes interconnected with directed links of unit capacity, where b_i stands for bit. This example shows that by encoding bits in node 3, two information units can be exchanged from source to destination at the same time. Network Coding can also achieve higher robustness by adopting distributive communications. Consider a relay system where the source node

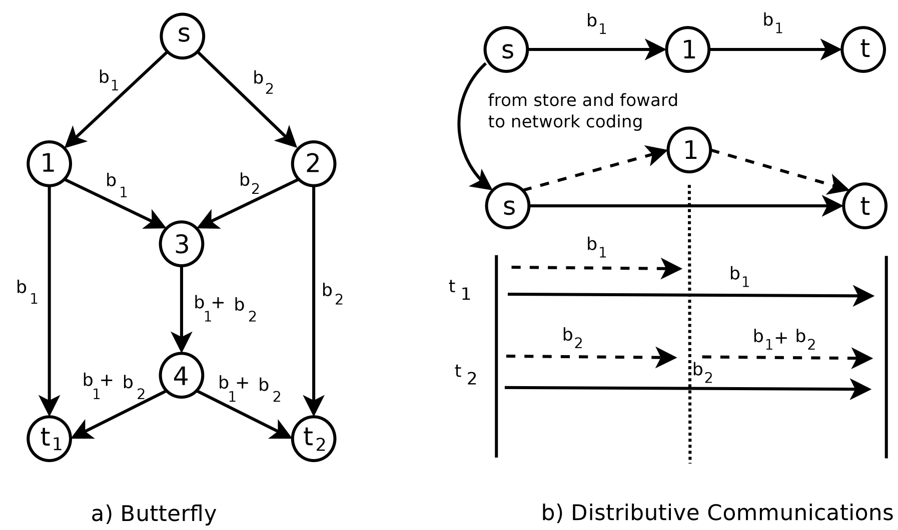


Figure 2.10: Throughput and robustness improvement examples with network coding.

sends a packet to an intermediate node that stores and forwards the information to the destination. With network coding the system can be changed so that the source sends information units to the relay as well as to the destination itself, reducing packet loss probabilities as seen in Figure 2.10 (b). The advantages of distributive networks with network coding can also be applied to reduce complexity [33]. Multicast sub-graphs, require Steiner tree computations [34] that can be simplified in certain scenarios by distributive systems with network coding. Finally, from a security point of view, network coding encodes packets that can only be decoded at the destination. This has two sides; from one side, discourages the spoofing of packets throughout the network since they are not decodable. From the other side, if an intermediate node is a malicious node, encoded packets will be delivered to the destination as source information, breaking the confidentiality of the information.

Considering the network coding evolution time line, although network coding has been attributed to Ahlswede *et al.* [32], Yeung and Zhang were first suggesting network coding features for satellite communications in 1998 [35]. Since then, network coding research in the field has been expanding mainly in encoding/decoding systems for a large number of nodes to achieve larger throughput values. Li, Yeung and Cai [36] found that linear encoding at the interior nodes is sufficient to achieve better performance. Koetter and Medard [37] expanded the work in [36] with linear encoder/decoder functions by finding polynomial coefficients and Ho *et al.* [38] improved this result by applying random linear codes for reliable communication. A more recent research approach proposed Physical Network Coding (PNC) [39] which has been widely researched with publications including [40] and [41].

The above work is based on theoretical and simulation results assuming combinations of multicast traffic, optimal scheduling and channel models with steady rates. Another approach to this method has shown a more practical angle; a widely known application is Avalanche [42], a peer to peer content distribution network. A similar suggestion was developed by Chung *et al.* who designed a peer to peer Streaming System named PNECOS that showed a reduction of bandwidth consumption for servers with network coding techniques [43]. Katti *et al.* presented COPE, a new architecture for wireless mesh networks, applicable to technologies such as IEEE 802.11 (Wi-Fi) and IEEE 802.16 (WiMax) [44]. Vingelman *et al.* developed a random linear network coding application based on OpenGl software that takes advantage of graphic cards to perform calculations related to network coding [45].

2.4.1 Network Coding Theory

In this subsection, network coding theory is explained by means of graphical examples. Single source single destination model is used to introduce network coding followed by single source multiple destinations models which leads to a wireless model utilised in our proposed design in Chapter 5.

2.4.1.1 Single Source Single Sink

A graph $G = (V, E)$ consists of a set of vertices, V, and a set of unidirectional connecting lines called edges, E. There are three type of vertices: the source , the intermediate node and the sink represented by s, a numbered circle and t respectively. Source node is essentially the transmitter of a communication system and does not receive any input edge. Intermediate nodes act as routing nodes and the sink is the receiver of the information generated by the transmitter. Nodes are connected by directional arrows that are defined by a numerical value, the capacity of the edge C. An edge is denoted as $e(i, j)$ with a direction from its "*initial point*" i to its "*terminal point*" j. The flow of an edge, f_{ij}, are the information units that are sent from node i to node j. In our graphs two conditions are imposed.

- The flow of an edge (i, j) in G is positive and equal or less than the capacity of the edge [46].
$$0 \leq f_{ij} \leq C_{ij} \tag{2.13}$$

- Each vertex i, excluding s and t, has an input flow the same as the output flow following the *conservation* law [46]. Inflow$(F_+(i))$ = Outflow$(F_-(i))$
$$F_+(i) = \sum_{k \in In(i)}^{n} f_{ki} \tag{2.14}$$
and
$$F_-(i) = \sum_{j \in Out(i)}^{n} f_{ij} \tag{2.15}$$

The conservation law requires that the output flow from s has to be equal to the input flow of the sink t which defines the flow of the graph, F. The goal is to maximise the flow of G given different sets of C_{ij}, in order to achieve the *MAX-flow*. F has to be then greater or equal than any capacity in the network.

A cut set is a set of edges of the network whereby the network is divided into two sets, one of them containing the source and the other the sink. The idea is to find out what is the flow of the graph by cutting the network because any flow from s to t must sometimes pass through some of theses edges. We denoted a cut set by (S, T) where S is a set of vertices that lies on the source side and T is a set of vertices that lies on the sink side. Figure 2.11 (a) shows an example of a single source single sink graph. The dotted line expresses a cut set on the graph. In this example $S = \{s, 1, 2\}$ and $T = \{t\}$. The capacity of a cut, with respect to the capacity of each edge, is defined as the sum of forward edges in the cut (S, T) and only the forward edges, where forward edges are defined as the edges of the direction from the source to the sink [46].

$$capacity(S, T) = \sum_{i \in S, j \in T} C_{ij} \tag{2.16}$$

A $MIN-cut$ is therefore a cut set where the capacity is less or equal than any other cut set of the graph. The illustration in Figure 2.11 (a) shows a minimum cut set of capacity 3. If the cut set is carried out in the same direction but above the nodes 1 and 2 the capacity is 4. Clearly the edges $(s, 1)$ and $(s, 2)$ with capacity 2 are actually not contributing in anything to the maximum flow of the graph because edges $(1, t)$ and $(2, t)$ can not handle more than 3 units of information per time. This intuitive example led to the *MAX-flow MIN-cut* theorem presented by Ford and Fulkerson at

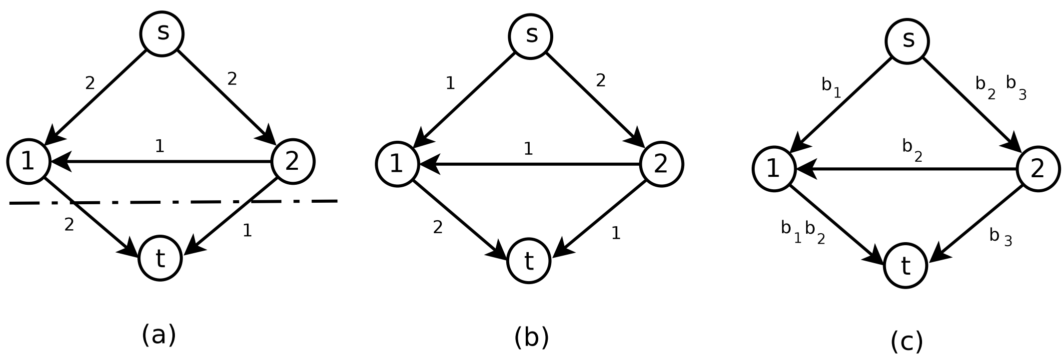


Figure 2.11: The single source single sink example. a) Capacities of the network, b) Max-flow network, c) Example with binary units to achieve max-flow [46].

the *Canadian Journal of Mathematics* [47].

Theorem 1. *The maximum flow in any network G is equal to the minimum cut set of the graph.*

In Figure 2.11 (a) an initial graph with different capacities is shown. The measurement of the cut sets proves that the minimum cut set is 3 and therefore the graph is adapted to Figure 2.11 (b). The exchange of information at the maximum rate is shown in Figure 2.11 (c) where 3 bits are sent from source to destination. It is important to point out that the delay introduced by each edge is considered as non existing. If other network topologies of single source single sink are studied it can be proved that the max-flow bound is always achieved [46].

2.4.1.2 Single Source Multiple Sinks

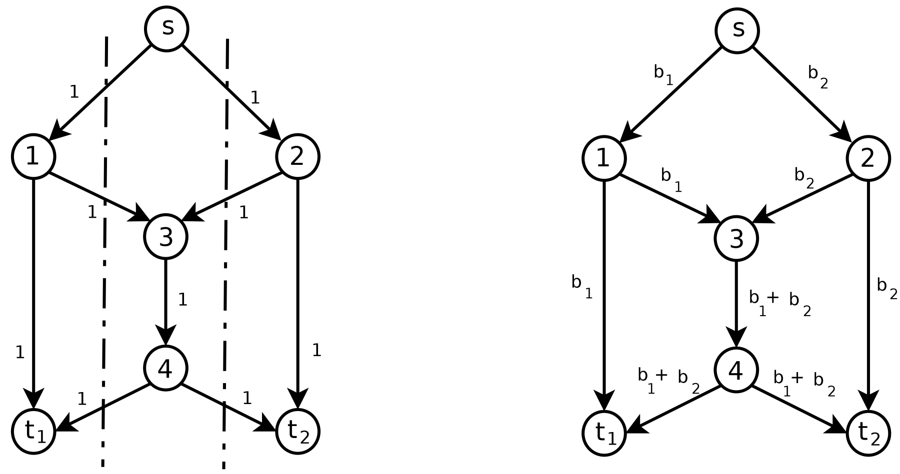
The butterfly graph example of Figure 2.12 shows an example of single source multiple sink graphs. In a graph G with sinks t_1,t_2, ... t_L , we denote the information rate received by each sink as γ. Hence we must have [46]

$$\gamma \leq F_{max}(t_l) \tag{2.17}$$

and

$$\gamma \leq min_l\{F_{max}(t_l)\} \tag{2.18}$$

where F_{max} is the maximum flow of a sink and equation (2.18) is the minimum flow of all maximum flows of the graph. This is explained in Figure 2.12. The first graph, Figure 2.12 (a), shows the capacity of the edges of the butterfly and the cuts required to compute the F_{max} of each sink. Recall that the max flow is the sum of the edge capacities going forward to the sink. It can be seen that $F_{max}(t_1) = 1+1 = 2$ and $F_{max}(t_2) = 2$ where the max flow of the graph is then $\gamma = 2$. Figure 2.12 (b) shows how to achieve such max flow. The source sends two information bits simultaneously from $(s, 1)$ and $(s, 2)$. b_1 and b_2 are replicated by nodes 1 and 2 respectively. At this point node 3 disposes of two bit of information but only one can be sent. Since the routing scheme for the sinks is fixed, node 3 can combine both bits and send them to node 4 where they are forwarded to the sinks. If no network coding schemes were



(a) (b)

Figure 2.12: The butterfly example is a single source two sink example. a) Capacities of the network, b) Max-flow example.

used, node 3 would have to send only one of the information units , either b_1 or b_2 and therefore both sinks would not have received all the information sent by the source.

The butterfly graph, a single source and multiple sink model, has shown that network coding is a necessary feature to achieve the *MAX-flow MIN-cut* theorem, however real applications might not look like the butterfly model. In a wireless broadcast network a central node called the access point (for instance a satellite or a hot spot of an IEEE 802.11 network) is responsible for broadcasting or unicasting information to the receivers/sinks. Using the single source multiple sinks approach next subsection describes the linear network coding problem.

2.4.1.3 Linear Network Codes

We present the linear network code problem by an acyclic network G, where the channels are assumed to have unit capacity of a symbol defined over a field $\mathbb{F}$ and parallel channels are allowed. There exists a unique source node s that transmits γ number of symbols per unit time that corresponds to the original message. The example in Figure 2.13 (a), has two imaginary input edges and represents an acyclic network. Each link has unit capacity and the max flow of the network can be computed with the use of cut sets as $\gamma = 2$. The routing scheme to achieve max flow is shown in Figure 2.13 (b). This graph is a good representation of a wireless network where the

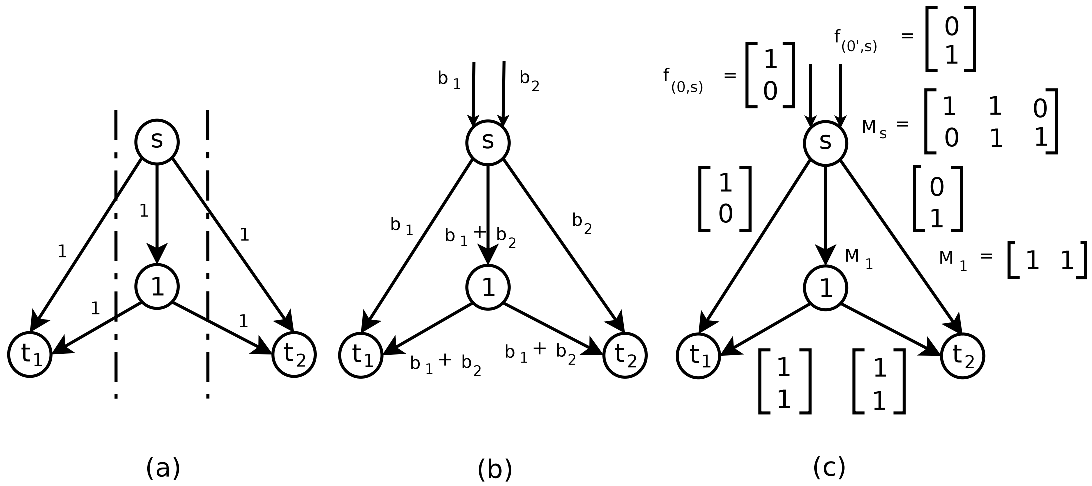


Figure 2.13: The acyclic network model for a wireless network. a) Capacities of the network, b) Max-flow example, c) The encoding system for a wireless communication model.

source node and node 1 are the central node, i.e. the access point, and t_1, t_2 are the receivers or users of the network.

Focusing on Figure 2.13 (c) where the wireless network graph has been disseminated to describe linear network coding. The source node in this case receives two flows, $f_{(0,s)}$ and $f_{(0',s)}$ that represent the source information $\vec{x}$ where $\vec{x} = X_d \cdot f_{d,s}$, $d \in In(s)$ and $X_d \in \mathbb{F}^{\gamma}$. The output of s is received by k nodes in the form of $(f_{s,j})_{i,k} \cdot M_{s,j}$, where $j \in Out(s)$, i and k represent the rows and columns of the output matrix $f_{s,j}$ and M is the mapping encoder of each node. Since the *conservation law* has to be accomplished, next hop symbol can be calculated as follows.

$$(f_{s,j})_{i,k} = \sum_{k \in Out(s)} \vec{x} \cdot M_{s,j} = \sum_{k \in Out(s)} (X_d \cdot f_{d,s}) \cdot M_{s,j} = \sum_{k \in Out(s)} X_d \cdot (f_{d,s} \cdot M_{s,j}) \quad (2.19)$$

In this way, the original message is transmitted from node to node and can be decoded as long as the mapping of each encoder, i.e. $M_{s,j}$ in this case, is known. In Figure 2.13 (c) there are two encoding nodes (s and node 1) and 7 flows. The two sources received at the source form a matrix such that

$$\begin{aligned} f_{d,s} &= \begin{bmatrix} f_{(0,s)} & f_{(0',s)} \end{bmatrix} = \begin{bmatrix} 1 & 0 \\ 0 & 1 \end{bmatrix} \\ X_d &= \begin{bmatrix} z_1 & z_2 \end{bmatrix}, \text{where } z_i \in \mathbb{F}^{\gamma} \end{aligned} \tag{2.20}$$

The encoders $M_{s,j}$ and $M_{1,l}$ are constructed by rows that define the input symbol and columns representing the next hop channel. Hence, $M_{s,j}$ with a two input node has two rows and since it is connected to 3 nodes it is constructed with 3 columns. The same construction method is applied to node 1 where there is one input symbol, thus one row, and two output channels giving two columns. The encoding system is associative, i.e. the order of the operations can be changed, thus the mapping system in Figure 2.13 (c) can be defined without considering the input symbols:

$$(f_{s,j})_{i,k} = f_{d,s} \cdot M_{s,j} = \begin{bmatrix} 1 & 0 \\ 0 & 1 \end{bmatrix} \times \begin{bmatrix} 1 & 1 & 0 \\ 0 & 1 & 1 \end{bmatrix} = \begin{bmatrix} 1 & 1 & 0 \\ 0 & 1 & 1 \end{bmatrix}, \tag{2.21}$$

$$f_{s,t_1} = (f_{s,j})_{i,1} = \begin{bmatrix} 1 \\ 0 \end{bmatrix}, \tag{2.22}$$

$$f_{s,1} = (f_{s,j})_{i,2} = \begin{bmatrix} 1 \\ 1 \end{bmatrix}, \tag{2.23}$$

$$f_{s,t_2} = (f_{s,j})_{i,3} = \begin{bmatrix} 0 \\ 1 \end{bmatrix}, \tag{2.24}$$

$$(f_{1,j})_{i,k} = f_{s,1} \times M_{1,l} = \begin{bmatrix} 1 \\ 1 \end{bmatrix} \times \begin{bmatrix} 1 & 1 \end{bmatrix} = \begin{bmatrix} 1 & 1 \\ 1 & 1 \end{bmatrix}, \tag{2.25}$$

$$f_{1,t_1} = (f_{1,j})_{i,1} = f_{1,t_2} = (f_{1,j})_{i,2} = \begin{bmatrix} 1 \\ 1 \end{bmatrix}, \tag{2.26}$$

On one hand, the encoder system is a set of linear equations where the symbols are defined at the field specified by the designer. On the other hand, the decoder of the system can be defined by a set of equations received at the sink. For instance, sink t_1 receives two flows, $f_{(s,t_1)}$ and $f_{(1,t_1)}$. Each row of received matrix flow is the combination of the initial message. Therefore the decoder has to solve the following equations

$$f_{(s,t_1)} \Rightarrow (f_{(s,j)})_{1,1} + (f_{(s,j)})_{2,1} = symbol_1 \tag{2.27}$$

$$f_{(1,t_1)} \Rightarrow (f_{(1,j)})_{1,1} + (f_{(1,j)})_{2,1} = symbol_2 \tag{2.28}$$

where + is the sum in the base field defined by the designer. The set of linear equations can be solved because $(f_{(s,j)})_{2,1} = 0$ and $(f_{(s,j)})_{1,1} = (f_{(1,j)})_{1,1}$. Thus equation (2.28) can be rewritten as $symbol_1 + (f_{(1,j)})_{2,1} = symbol_2$ and solved as

$$(f_{(1,j)})_{2,1} = symbol_2 - symbol_1 \tag{2.29}$$

Accordingly, to decode the information in sink t_2 the step process is the same as sink t_1.

Example: Let us now consider that the source wants to send two symbols z_1 and z_2 defined over a field $\mathbb{F}_2 = \{0, 1\}$. The source will then receive two information units defined as $X_d \cdot f_{(d,s)}$ which produces a matrix such that

$$\vec{x} = X_d \cdot f_{d,s} = \begin{bmatrix} z_1 & z_2 \end{bmatrix} \cdot \begin{bmatrix} 1 & 0 \\ 0 & 1 \end{bmatrix} = \begin{bmatrix} z_1 & z_2 \end{bmatrix} \tag{2.30}$$

the output flow of the source node is calculated using equation (2.19) and the output of node 1 using the mapping in equation (2.25).

$$(f_{s,j})_{i,k} = \vec{x} \cdot M_{s,j} = \begin{bmatrix} z_1 & z_2 \end{bmatrix} \times \begin{bmatrix} 1 & 1 & 0 \\ 0 & 1 & 1 \end{bmatrix} = \begin{bmatrix} z_1 & z_1 + z_2 & z_2 \end{bmatrix} \tag{2.31}$$

$$(f_{1,j})_{i,k} = f_{s,1} \cdot M_{1,l} = \begin{bmatrix} z_1 + z_2 \end{bmatrix} \times \begin{bmatrix} 1 & 1 \end{bmatrix} = \begin{bmatrix} z_1 + z_2 & z_1 + z_2 \end{bmatrix} \tag{2.32}$$

using equations (2.27) and (2.28) the set of equations that are solved for t_1 are

$$\begin{aligned} symbol_1 &= f_{s,t_1} = z_1 \\ symbol_2 &= f_{1,t_1} = (z_1 + z_2) \end{aligned} \tag{2.33}$$

solving the set of equations as

$$symbol_2 = (z_1 + z_2) - z_1 = (z_1 + z_2) + z_1 = z_2, \text{from } 2.33 \tag{2.34}$$

where minus sign is substituted by a sum because in $\mathbb{F}_2$ $+ = -$. Equally the set of equations to be solved by sink t_2 are

$$\begin{aligned} symbol_1 &= f_{s,t_2} = z_2 \\ symbol_2 &= f_{1,t_2} = (z_1 + z_2) \end{aligned} \tag{2.35}$$

solving the set of equations as

$$symbol_2 = (z_1 + z_2) - z_2 = (z_1 + z_2) + z_2 = z_1, \text{from } 2.35 \tag{2.36}$$

Hence, if the mapping system is known, i.e. $M_{s,j}$ and $M_{1,l}$, each sink can decode sent information. This graph (see Figure 2.13 (c)) is a representation itself of a wireless broadcast network. On one hand, the base station offering a connection to the users is equivalent to nodes s and 1 where received information or uplinks are the inputs to node s and the sent information or downlinks are the outputs of node 1. On the other hand, the users are represented by the sinks and the set of edges that connect them to the source, i.e. $e(s, t_i)$. In other words, $f_{0,s}$ and $f_{0',s}$ are known to t_1 and t_2 respectively. In a real scenario this means that a user of a BSS has a buffer of sent information allowing the receivers to decode incoming packets from node 1. This is the key feature to understand network coding over wireless networks.

Consider that the encoder matrices are known by both, the end user and the access point. An end user sends bits of information to the access point and keeps a buffer memory with what has been sent. This means that end user has either z_1 or z_2 stored. At this point the access point is aware that the end node is storing sent information and proceeds to encode received information before transmitting it to all users. The end nodes have stored sent information which means equation (2.27) is already at the receiver. If the access point sends the encoded message in equation (2.28) the user can retrieve the information by solving the system. The procedure guarantees a remarkable gain in throughput but there is still a fact that has not been considered so far, the delay. The graphs presented here have an ideal propagation feature free of delay. In reality, delay can restrict a service where if exceeds a maximum value might not be feasible to perform. Since VoIP is one of these services, the study of its performance is analysed over a wireless network such as IEEE 802.11b in Chapter 5.

2.5 Conclusion

In this chapter, SIP/SDP and RTP/RTCP are described to understand the signalling system and audio exchange procedures within a VoIP network. State of the art audio codecs G.711, GSM 06.10 and Speex are chosen because they are a good representation from the variety of codecs currently available. VoIP phone calls are assessed throughout this thesis with the MOS and the E-model as presented in this chapter. The speech prediction model is followed by a network coding introduction that gives an insight to the design of these network architectures. The study of heterogeneous network performance is addressed in the following chapter with a MOS test for an active SIP proxy GW.

CHAPTER 3

VoIP Quality in Heterogeneous Networks

3.1 Introduction

NOWADAYS due to the multiplicity of network communication solutions there is an increased possibility that VoIP calls not only take part within a single network but rather within a number of networks that often function using different standards and link layers. SIP has been widely adopted as the signalling system for PSN and thus the integration with existing technologies is important in order to offer full services. From an end-to-end user perspective SIP GWs are SIP entities that cross interact with these technologies and their performance can become critical. Consider that *Alice* uses a soft phone at her office desktop to call *Bob* whose location is unknown to *Alice*, as shown in Figure 1.1 on page 24. In an ideal scenario, *Alice* would call the UA corresponding to *Bob* and so the task that the network has to proceed is to locate *Bob*. Regardless of *Bob* using a land line, a cell phone or an office computer, *Alice* expects good phone call quality. Therefore, the critical subject is to know what are the requirements from the network's perspective to achieve such transparency.

Firstly, the signalling systems from the differing technologies must be translated to guarantee that the location and availability of *Bob* can be determined. Secondly, voice data is exchanged between end nodes through the GW, but the quality degradation of a single network can affect the whole conversation. For example, the fact that *Bob* may be on a mobile network which guarantees a minimum QoS will not be sufficient if *Alice*'s network is overloaded (consider *example A* at Figure 1.1 on page 24). This could eventually lead to a drop in a number of packets during the conversation. Heterogeneous network communications require that gateways not only provide simple forwarding methods but also end-to-end VoIP quality performance. Furthermore, if a SIP GW can sense the end-to-end speech quality then it can enforce entities to use methods such as Forward Error Correction (FEC) to overcome the situation.

The development of SIP entities have been growing in the last decade. Hyun et al. presented a SIP implementation and design of proxy and registrar in [48]. In [49] Huy et al. described User Agent (UA) implementation for voice services including call establishment, termination, registration and capability of negotiation. Other implementations take advantage of Private Branch Exchange (PBX) to develop an integrated telecommunication system. A recognised approach was achieved by Anjum et al. who developed ChaiTime [50], a Java Telephony API along with SIP to design an integrated system. Asterisk [51], an open source software implementation, is another example of a PBX that offers multiple protocol integration such as H.323, SIP and self developed IAX [52]. Although the developments above achieve different technological integration none of them considers the VoIP communication quality performance as a whole.

In this chapter, a SIP proxy emerged with a Media Gateway is presented as a test-bed implementation system to bridge IEEE 802.3 (wired) to GSM networks (wireless) and vice-versa. The platform senses in real-time the quality delivered to end users using the packet loss rate and supports FEC to overcome conditional scenarios where the wire section shows low end-to-end VoIP quality performance. This development is based on an embedded platform to provide the system with higher mobility and integration for future implementations. The comparison of the passive GW versus the active GW, i.e. with FEC system integrated, results in a better performance for active nodes according to the MOS rates.

The remainder of this chapter is organized as follows. Section 3.2 describes FEC for VoIP with an overview to channel estimations. Section 3.3 presents implementation tools and software architecture. Section 3.4, discusses the Mean Opinion Score (MOS) test results, and finally, Section 3.5 concludes the chapter.

3.2 FEC for VoIP

FEC for VoIP services was initially presented by Bolot [53] proving that, even at high packet loss probabilities, speech quality is maintained satisfactorily. More recently, advanced FEC methods have been tested for VoIP services; Wenyu *et al.* showed the feasibility of FEC under bursty loss Gilbert Elliot Model [54]. Sheng *et al.* presented a combination of FEC mechanism for wireless methods, including iterative decoders such as concatenated zigzag (CZ) [55]. Although most of this research focuses on FEC for VoIP they do not consider the whole communication system with a combination of wired and wireless networks, i.e. heterogeneous networks. In our integrated SIP server GW with FEC, the results in [53] are corroborated and its usage is extended to a wire-to-wireless network application. In addition, our application includes an integrated Markov Chain theory based loss model for test purposes; this allows us to emulate the behaviour of a multiplicity of channels.

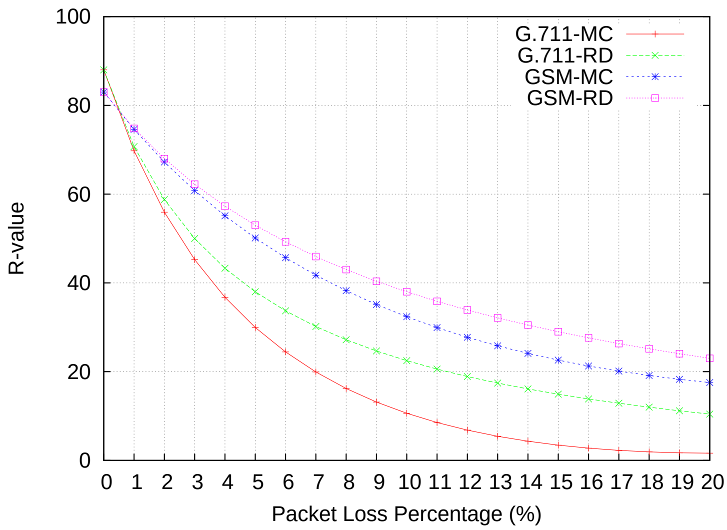


Figure 3.1: Speech quality prediction for a RanDomly interpreted channel (RD) versus a Markov Chain (MC) based model with codec parameters defined as $Bpl = 4.3$, $I_e = 0$ for G.711 and $Bpl = 39, 32, I_e = 24.3$ for GSM as specified in Chapter 4. FSMC is defined with $q = 0.7$.

3.2.1 Channel estimation with Markov Chain Theory

From a channel estimation point of view, channels are often modelled based on Bit Error Rates (BER) that consequently lead to frame errors. From a network point of view, channels are often modelled as Packet Error Rates (PER) or packet loss percentage. In Section 2.2.2 on Page 35 it is mentioned that RTCP exchanges the value of packet loss rate in relation to the link between end users. This percentage can have different interpretations, but mainly there are two schemes to follow. The packet loss percentage is either random or non-random. In case of a non random process, different mathematical models can be applied, where a widely applied model is Finite State Markov Chain (FSMC) models. Figure 3.1 represents the difference of both interpretations using the E-model for G.711 and GSM. From the illustration it is shown that the performance of Markov Chain (MC) based models has worse performance than RanDom (RD) based interpretation, reaching to a conclusion that the use of random packet loss model can be misleading in VoIP quality performance prediction.

FSMC are channel loss models often utilised in network modelling for wired and wireless communications. In wireless communications, FSMC models have been used

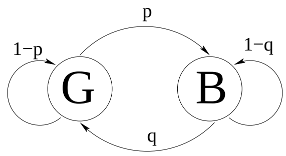


Figure 3.2: Fist-Order Finite State Markov Chain model [62].

to model error burst in fading channels, facilitating designers to analyse and improve performance measurements in both physical and medium access control services. In [56] it was proven that higher information rates can be achieved if FSMC memory is taken into account. A common approximation for fading channels is Rayleigh Probability Distribution Function (PDF). This PDF is divided into Markovian states depending on the nature of the fading channel [57], i.e. the higher the number of states the greater the complexity. Although higher order FSMC models have proven better in performance, they considerably increase the complexity which for a real-time channel modelling is not desirable. Hence in this thesis first order FSMC models are used due to their computational simplicity and accuracy [58].

In wired communications performance of the Internet is affected by routing schemes. Packet loss due to congestions in routers, delay or priority policies are every-day constraints. In [59–61] it is shown that the Internet packet loss can be represented as *on* and *off* states, whereby *on* refers to all packets being dropped and *off* signifies no packets are lost. This behaviour is well suited for a first order FSMC.

A single first order Markov Chain is shown in Figure 3.2 where $P(G- > B) = p$ represents the probability of moving from the Good state to the Bad State and $P(B- > G) = q$ the probability of moving from the Bad state to the Good.

If the channel is in the Good state packets are delivered to their destination whereas within the Bad state packets are dropped. The average probability of being in each state is then given by equation (3.1), where π_0 and π_1 are the Good and Bad states respectively [62].

$$\pi_0 = (q/(p+q)), \pi_1 = (p/(p+q)) \tag{3.1}$$

Figure 3.3 illustrates the packet loss probability for packets lost in a row with $q = 0.7$. The graph demonstrates that the probability of loosing the first packet is always the highest provided that previous packet was not lost. Consequently, the probability of loosing yet another packet in a row diminishes as the number of

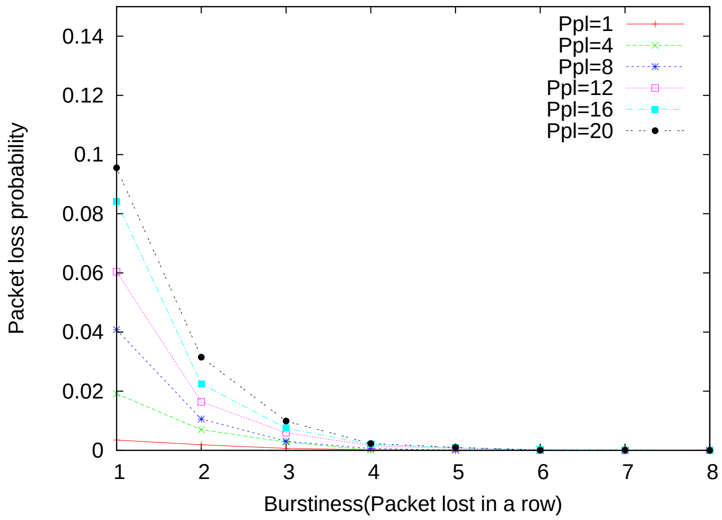


Figure 3.3: 2-state-Markov Chain loss model, q=0.7.

packets lost in a row increases. This graph clearly shows that if the UAs are able to avoid loosing a single packet the highest probability of packet loss can be decreased considerably and hence the perception of speech quality is higher. A method to avoid single packet losses is FEC with piggy backing as explained in the following subsection.

3.2.2 FEC Algorithms

FEC algorithms use redundant information to guarantee the quality of a VoIP session is not degraded. The aim of FEC utilisation in our system is to avoid single packet losses because FSMC showed a very high probability of loosing such packets.

There are a variety of algorithms for FEC illustrated in Figure 3.4 where RTP packets are represented by a Header (HDR) and a Payload (PL). FEC with repetition, as presented in Figure 3.4 (a), guarantees that if a single packet is lost it can be recovered by the next incoming message. Although this is a practical approach it introduces n number of redundant packets where n is considered as the total number of packets sent in one RTP stream. The following case shown in Figure 3.4 (b), refers to a method that follows the repetition algorithm but in this case is carried out by padding the current message in the following packet. This method sends $n-1$ redundant packets, however the number of headers required for this method is reduced to the half, resulting in a more efficient option. The last option, in Figure 3.4 (c),

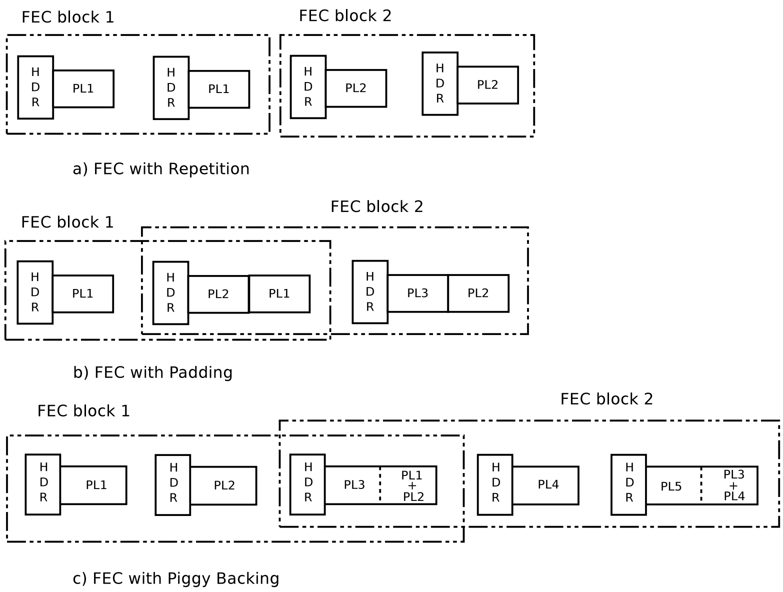


Figure 3.4: Different Forward Error Correction algorithms where HDR stands for header and PL payload.

depicts a FEC system where packets are *xor*-ed and padded to the third payload. This method also named as FEC with piggy backing, ensures that a single packet is not lost with a probability of [54]

$$\pi_1.FEC = \frac{p}{(p+q)}((1-q)^2 + pq + q(1-q)) \tag{3.2}$$

where the equation is constructed by starting at the Bad state, i.e. one packet is already lost, and multiplying it with the addition of three possible cases where recovering lost packets is not possible; the first case refers to loosing the following two packets, $(1-q)^2$; the second represents the case when the second packet is received but the third is lost, pq and finally the last possibility is to loss the second packet but receive the third one, $(1-q)p$. This method only requires $(n-1)/2$ number of redundant packets, which is a considerable gain in comparison to the other two options. For this reason, the following SIP proxy GW is implemented using FEC with piggy backing.

3.3 System Implementation

In this chapter, an embedded SIP proxy GW connected to a GSM capable of working on a Small Office/Home Office (SOHO) environment written in C code is presented. The implementation is based on a general purpose board, a GSM modem and an expansion board, specifically designed for this implementation.

The test board is based on NGW100, a general purpose development board for the AT32AP7000 processor [63]. The 32-bit Micro-Controller Unit (MCU) is a processor core with an architecture created specifically for cost-sensitive embedded applications that require both high performance and low power consumption. This architecture is based on a RISC processor core designed to do more processing per clock cycle so that the same throughput can be achieved at a lower clock frequency with substantially less power consumption. The NGW100 is equipped with a 8MB Parallel flash, a non-volatile flash memory reloaded with the U-Boot boot loader for Linux, 8MB Serial flash dedicated to store permanent code and 32MB SDRAM used to store temporary data. The expansion board has a CS4202 mixed-signal serial audio codec with an integrated headphone power amplifier compliant with the Intel Audio Codec 97 specification, referred as AC 97 capable of supporting PCM. Finally, the GSM modem is controlled by a level converter from a serial port from the NGW100 and its audio input/output are connected to the expansion board. Telit has been chosen for this development [64]. The embedded SIP server incorporates Linux 2.6.23 Operation System (OS) and this can be controlled by both serial port and Telnet. Vendors cross compiler and Integrated Design Environment (IDE) has been used in a Fedora Core 8 to develop the software. Further description of the implementation can be found at Appendix B .

Software implementation is based on three main sections: Firstly the SIP server Gateway's functionality and the call flow of a VoIP session is described, secondly, our algorithm to support FEC with *piggy backing* (encoder/decoder) is presented and thirdly, state machines synchronisation for the SIP and Modem network is explained.

3.3.0.1 Embedded SIP Server merged with a Media Gateway

The Embedded SIP server GW has two main tasks; on one hand it must control the signalling system from both VoIP and Mobile network, and on the other hand it must handle the RTP protocol and translate it to the GSM network audio input. oRtp [65]

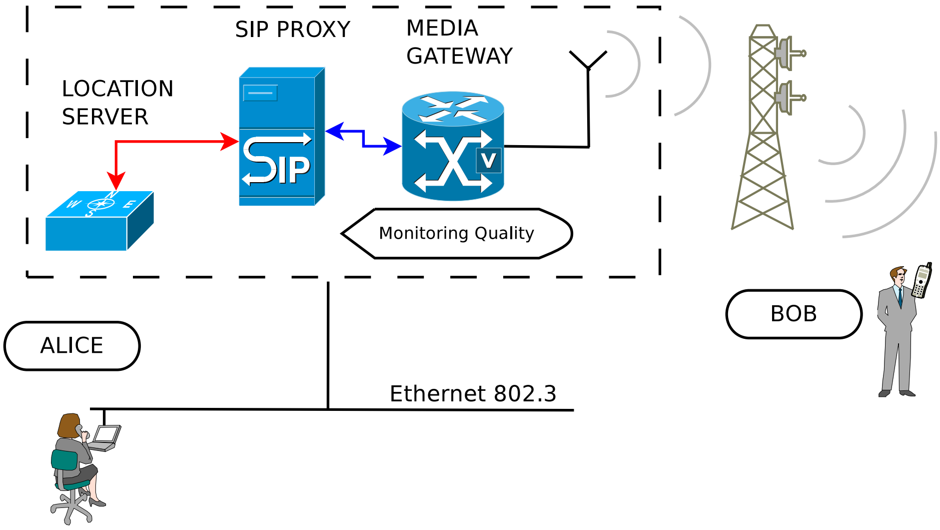


Figure 3.5: Block diagram of the SIP server merged with a Media Gateway connected to a mobile network.

and oSIP [66] libraries are used to implement the SIP server. The block diagram of the implementation is presented in Figure 3.5. The figure shows the set up for VoIP calls carried out in the experiments where Alice is connected to the SIP proxy merged with the Media Gateway. The gateway includes a monitoring system that allows to the entity to react under poor quality circumstances. The media gateway is connected to the mobile network through a modem that provides a communication system from Alice to Bob through the base station. The flow chart of a current VoIP call is presented next.

Figure 3.6 shows the signalling flow of the embedded SIP server merged with a Media Gateway. Several assumptions are made for simplicity; Mobile Stations (MS) have already been assigned their channel number to guarantee synchronisation with the Base Station (BS) and Authentication and Ciphering processes have been discarded. In the graph the Mobile Control Centre (MCC) is a unified concept that includes; Base Station Controller (BSC), Mobile Switching Centre (MSC), Visitor Location Register (VLR), Home Location Register (HLR) and Authentication Centre (AuC). BSC is in charge of several BS, MSC is responsible for switching routing and controlling a call, VLR stores all relevant information related to the user, HLR is responsible for the user location information and the AUC is in control of authentication. Readers are referred to [67] for further details.

This flow chart starts with the registration from *Alice*'s UA to the Embedded SIP

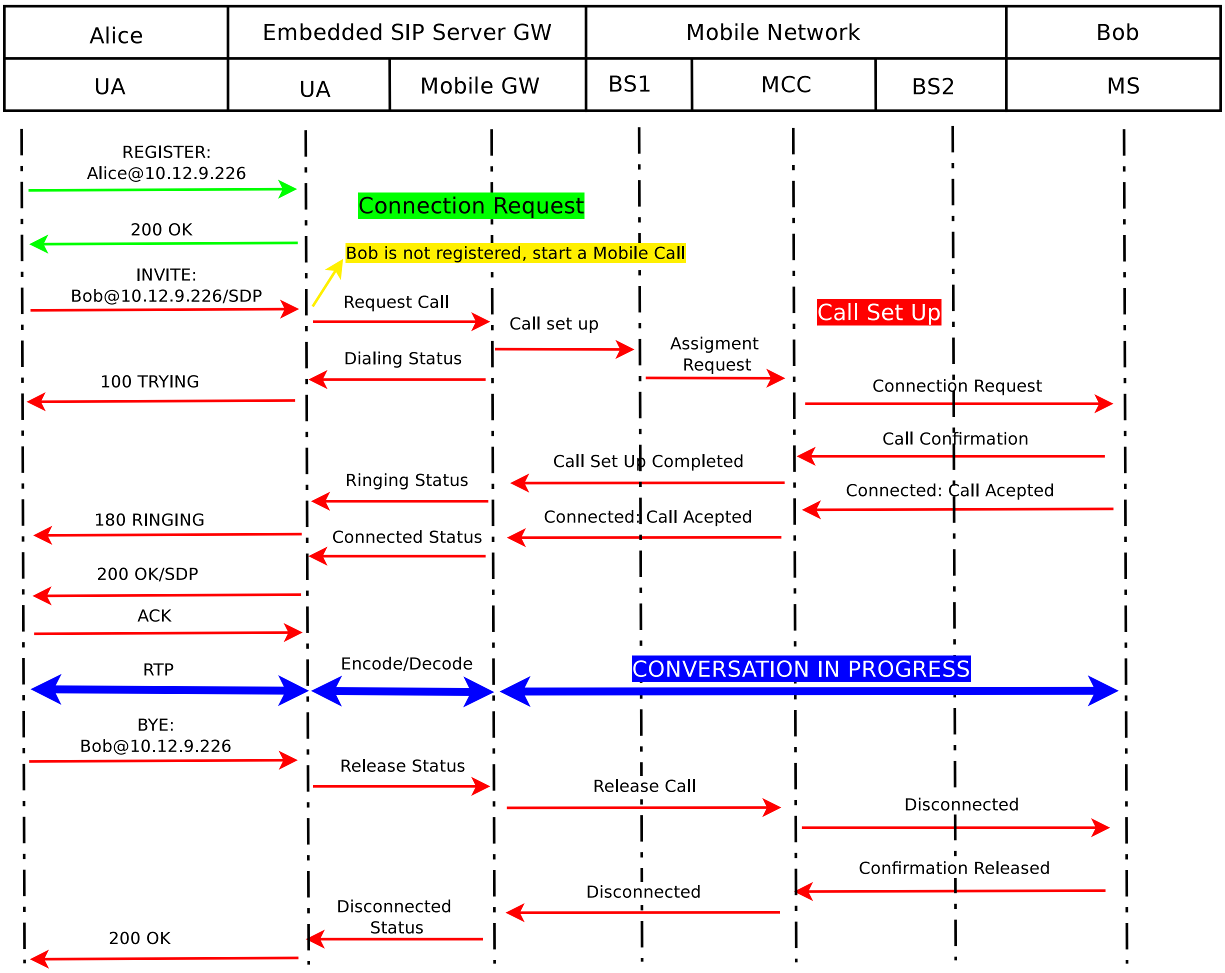


Figure 3.6: Flow chart for a wire to mobile network VoIP call [68].

Server GW. If this person has permission to be registered, the server will reply with a *200 OK* accepting the request or it will be rejected. *Alice* generates a phone call to *Bob* by sending an *Invite* request including its options for audio codecs, i.e. SDP. The embedded SIP server checks its Register Servers entrance and codec compatibility. *Bob* is not in the local area network and therefore a mobile network session is initiated by sending a request call to the Mobile Gateway (Mobile GW), see Figure 3.6. The Mobile GW is connected through the Base Station 1 (BS1) to the MCC and forwards a Call request to contact *Bob*. MCC checks in the routing tables to track down *Bob*'s Mobile Station. *Bob*'s MS receives a Connection Request while the Mobile GW sends a *100 Trying* response to advise *Alice*'s UA that the call has been processed. *Bob*'s MS confirms that the call has gone through and starts ringing and in return the

Embedded Server SIP GW sends a *180 Ringing* to *Alice*. If *Bob*'s phone is switched off or not available, then this will be noticed by the Embedded Server to *Alice* and the phone call will be finished. In this case, *Bob* accepts the phone call and a *200 OK* message is sent to *Alice* with codec priorities. It is after this process when *Alice* and *Bob* can talk through the Embedded SIP Server GW. The conversation is finished in this example by *Alice* sending a *Bye* request. This request is forwarded to the MCC and *Bob* is informed that *Alice* has hung the phone up. Confirmation is received at the embedded Server and a response with a *200 OK* is sent.

The embedded SIP server GW allows a routing scheme to track down the user location. If the user is not logged in, the phone call is routed to the mobile network. This internal routing system is a very useful feature that always makes sure a cheaper phone call will be performed, avoiding any extra action from the end user and blindly integrating both technologies.

As mentioned in the introduction of this chapter, it is possible that *Alice*'s side (wired communication) might not be able to deliver packets degrading the VoIP performance. This problem can be tackled with the use of FEC algorithms as shown next.

3.3.0.2 FEC with piggy backing algorithm

FEC with piggy backing improves the quality of an end-to-end VoIP session. In this chapter, the SIP server GW is presented with the support of FEC with piggy backing. The GW becomes an active node within the network that sensing the QoS of the communication can enforce the use of FEC.

Both FEC encoder and decoder have been developed in C programming language at the wire network section. The encoder block code algorithm for Figure 3.4 (c) is presented in Figure 3.7. Once all variables are set to their default values, the algorithm starts reading voice frames from the Analogue to Digital (A/D) converter, $Praw$ and encodes them in G.711 audio codec, Pn. If this is the first packet then $Pfirst = 1$ and the packet is sent using conventional method after being copied to an internal buffer ($cpy(Pn, Pa)$). At the time of reading the second packet, $Pfirst = 2$ the packet is sent conventionally as well as being copied to a different internal buffer ($cpy(Pn, Pb)$). Third packet's case is different because $Pfirst = 3$. Therefore the xor of previous two packets is performed and the output is concatenated with the original message, i.e. $Pxor = xor(Pa, Pb)$ and $Pfec = conc(Pn, Pxor)$. The new

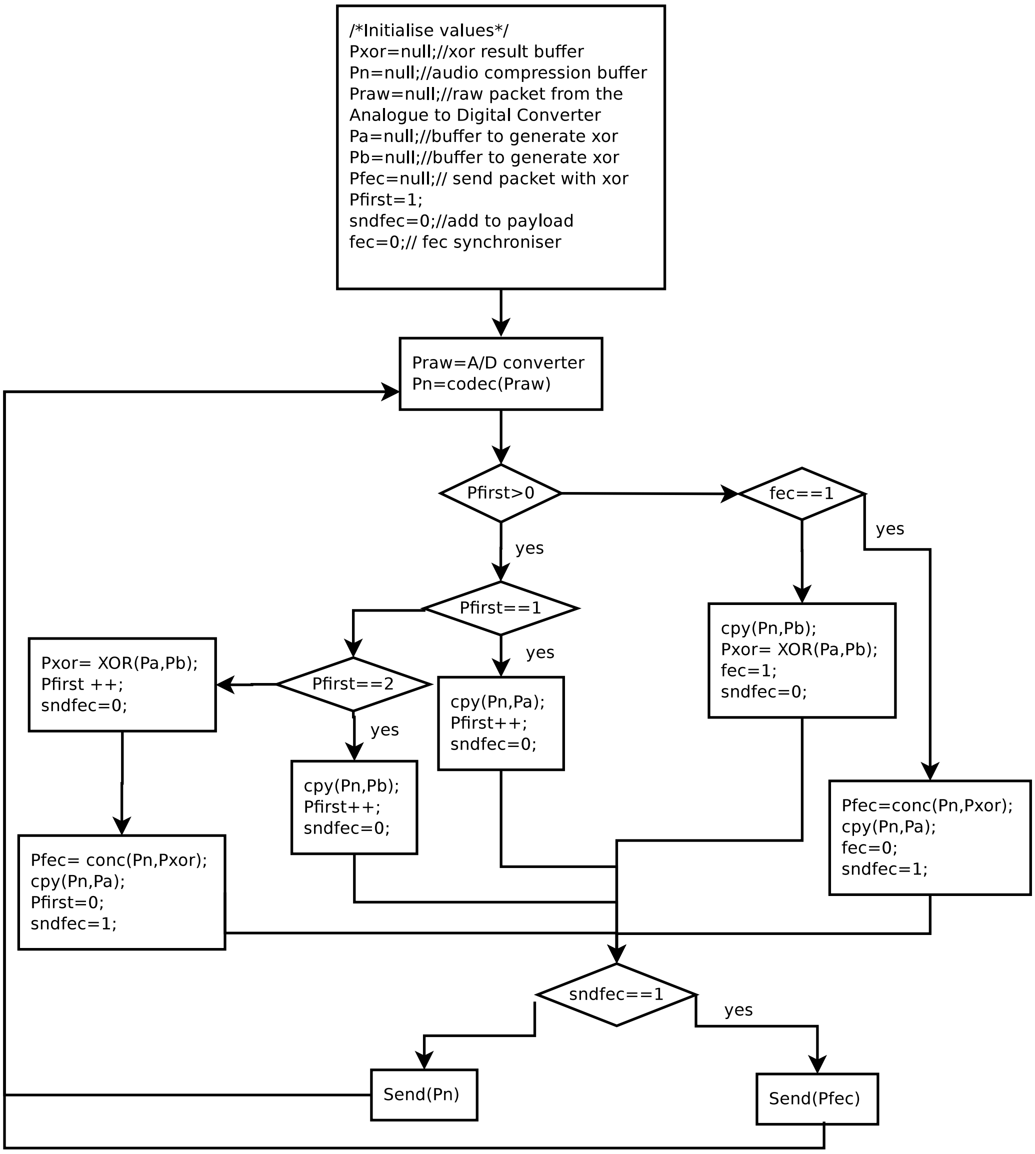


Figure 3.7: FEC encoder with piggy backing [68].

packet is sent with the FEC padding and the third packet is copied to the internal buffer Pa. Since the variable $Pfirst$ is set to zero the forth packet takes diagram's right tree in $Pfirst > 0$. In this case $fec = 0$ and hence, forth packet is sent without FEC but it is copied to the internal buffer Pb. Consequently the fifth packet follows the path of the forth packet but this time FEC is added entering in a loop that sends packets with FEC as piggy backing.

The decoder reads the sequence number of received packets, decides whether a packet is lost or not and if possible implements the *xor* to recover the lost packet. The pseudo code for this algorithm is presented in Algorithm 3.1. Received packets are divided into a set of three packets called *fecblock*s as shown in Figure 3.4 c), so that synchronisation within received packets is not lost. A socket is the source of received packets which are copied to an internal buffer, Prx, where the payload and the sequence number are stored in $Prx.buf$ and $Prx.seq$, respectively. Consider that the first FEC block does not loose any packet; Since the first received packet opens a new FEC block, it is copied by the ADD function to the FEC buffer. This function takes the payload and sequence number from Prx and copies them to $Pfec.buf$ and $Pfec.seq$. $Pfec$ is considered as an auxiliary buffer to perform FEC when necessary. Once the first packet of the block is received the second packet arrives to the socket. In this case, the sequence number of the received packet is compared with the one in $Pfec$ and stored as $diff = Packet2 - Packet1 = 1$. Because the received packet does not contain FEC, the program jumps from line 8 to line 26 where $diff$ value is checked to be equal or greater than two. If the latter occurs it means that several packet looses have happened. Since this is not the case, the payload retained at the FEC buffer and the received payload are copied to the jitter buffer in this order by using the $ADDtoJitter$ function. Note that the order of adding the payloads to the jitter buffer have to be respected. Before closing the loop, the initiation of a new FEC block is allowed in line 32. Consequently the third packet received is stored at $Pfec$. Consider that the new FEC block losses the forth packet, which means that the following received packet is the fifth which contains a FEC padded payload. Line 8 is then $true$ and the difference between sequence numbers is considered. In this case, because packet four is lost, $diff = 5 - 3 = 2$ and the recovery of the packet is performed with the third and the fifth packets, namely $Pnew$ (see line 10). At this point, there are two possibilities: the first or the second packet of a FEC block is lost, which means that $diff = 1$ or $diff = 2$, respectively.

It is important to determine the order in which payloads are copied to the jitter buffer. In our case the second packet is lost and therefore FEC buffer, i.e. packet three, is copied before $Pnew$. There is also the case that several packets have been lost having no chance of packet recovery. This is reflected in line 22 where only the FEC buffer is copied to the jitter buffer. The system continuous the loop until the socket is closed.

Algorithm 3.1 FEC Decoder with piggy backing [68].

```
newfecblock = 1
▷ Synchronisation
while Prx = packet received do
  if newfecblock then
    ADD(Prx to Pfec)
    newfecblock = 0
  else
    diff = Prx.seq − Pfec.seq
    if Prx with FEC == (True) then
      if diff ≤ 2 then
        Pnew = XOR(Prx.buf, Pfec.buf)
        ▷ Recover Lost packet
        if diff == 2 then
          ADDtoJitter(Pfec.buf)
          ADDtoJitter(Pnew.buf)
        else
          ADDtoJitter(Pnew.buf)
          ADDtoJitter(Pfec.buf)
        end if
        ADD (Prx to Pfec)
      else
        ▷ Several packets lost
        ADDtoJitter(Pfec.buf)
        ADD (Prx to Pfec)
      end if
    else
      if diff ≥ 2 then
        ADDtoJitter(Pfec.buf)
        ADD (Prx to Pfec)
      else
        ADDtoJitter(Pfec.buf)
        ADDtoJitter(Prx.buf)
        newfecblock = 1
      end if
    end if
  end if
end while
```

3.3.0.3 Synchronisation

The software uses dynamically allocated threads to handle both the signalling system and audio packets. There are two audio threads: RTP packet receiver and RTP packet sender. Similarly, the signalling system is comprised of two threads *SIP_thread* and *Modem_thread*.

Figure 3.8 depicts the finite state machine for the *SIP_thread* responsible for incoming calls (*SIP_fsm*) whereas Figure 3.9 shows the modem finite state machine (*GSM_fsm*). The former's initial status is at *Invite Received* and the latter's is at *Checking Status*. When a call request is received and a mobile phone call is initiated, codec type and modem availability are checked before *SIP_fsm* changes to *Trying*. With regards to the modem thread, *Invite* is a call event that changes the status to *Dialling* provided that the modem is not in use by any other previous thread. If no other event occurs during this process, *GSM_fsm* changes to *Ringing Status* and subsequently makes the *SIP_fsm's* status shift to *Ringing*. At this stage, the mobile station is ringing. If the end user accepts the phone call the *GSM_fsm* and the *SIP_fsm* switch to connected status. The conversation can be finished by any of the two end users returning back to the *Checking Status* and *Terminated* for *GSM_fsm* and *SIP_fsm* respectively. The colours of Figure 3.8 and 3.9 illustrate the follow up of the status of each synchronisation state-machine.

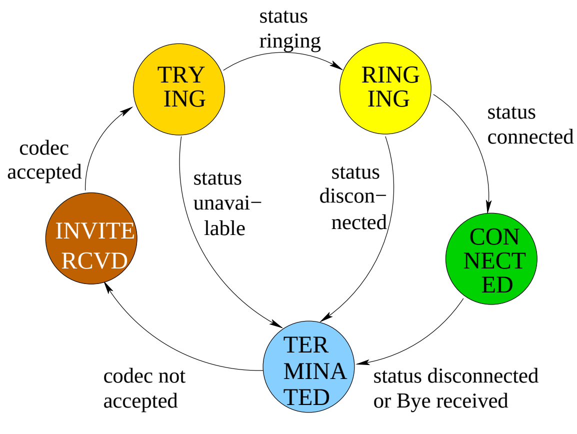


Figure 3.8: SIP thread state machine [68].

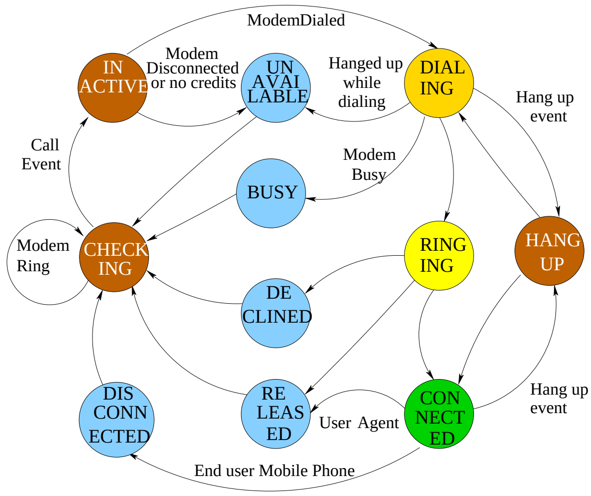


Figure 3.9: Modem thread state machine [68].

3.4 Testing and Results

The embedded SIP server GW has been successfully tested with commercial softphones such as X-LITE [69], SjPhone [70] as well as open source Ekiga [71]. None of these sofphones offer FEC with piggy backing and therefore a self-developed User Agent with FEC is utilised in these experiments.

The test bench has a first order FSMC integrated for both audio threads (sender and receiver) with $q = 0.7$ following tested values for an Internet communication record cited in [53]. The packet losses were set up from 0 to 20% and G.711 [19] was chosen as a codec to the wired channel. Speech at the softphone was recorded to evaluate its quality according to the MOS [1] recommendation (see Section 2.3.2 on Page 45).

The experiment consists of a set of VoIP phone calls from a computer to a mobile phone where packet loss rates at the SIP gateway vary from 0 to 20 %. Firstly, phone calls without FEC algorithm are performed and secondly, FEC with *piggy backing* algorithm are considered. In addition, single network audio codec performance is also added to these experiments to prove that heterogeneous networks performance is worse than single network communication. The methodology followed for these results is based on the simulation tool developed in this thesis and explained further in Section 4.2 on Page 84. In Figure 3.10 the MOS test average results are illustrated depending on the packet loss percentage applied to the wired channel. These exper-

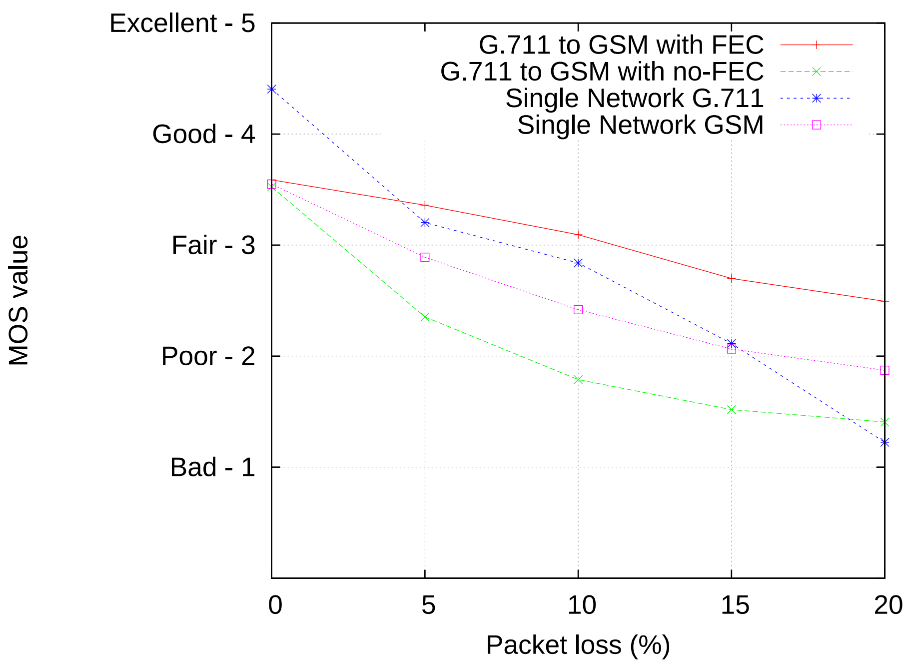


Figure 3.10: MOS test with q=0.7 for wire-to-wireless connection and single network connection [68].

iments have been assessed by over 20 people, in English, with both native and non native listeners following the MOS standard requirements.

On one hand, the illustration shows *FEC* and *no-FEC* sessions perform equally when the channel is free of packet losses. As the packet loss rate increases, FEC's influence is more noticeable. This reveals that with a 10% of packet losses, FEC registers a performance of a fairly good connection whereas the no-FEC channel is evaluated as worse than poor. Even when considering a 20% packet loss rate, FEC shows a poor performance but still achieves adequate speech quality. On the other hand, single network VoIP performance for G.711 and GSM 06.10 is depicted. The comparison between cascaded two network performance and single network performance highlights the impact of dual codec connection and corroborates that the awareness of cascaded two networks connection can be advantageous to improve over all performance. Consider the case of five percent of packet loss; a single network performance for G.711 is 3.2 whereas double channel connection, i.e. *G.711 to GSM with no-FEC*, shows a 2.3. This difference means that if a node is not aware that there is more than one audio codec taking part in the communication, it could believe that overall performance value is fair, i.e. 3.2, in stead of poor, i.e. 2.3. Hence, if intermediate

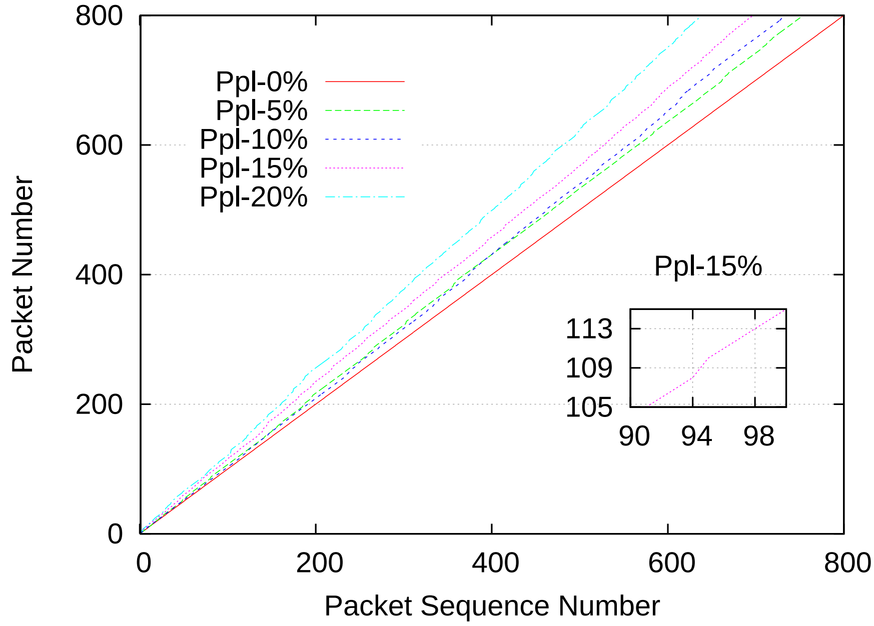


Figure 3.11: Received VoIP packet sequence result.
Received VoIP packet sequence result.

nodes are aware of the communication system their assessment of the quality is more accurate and the decision of considering FEC methods is more accurate.

One of the drawbacks related to FEC in addition to the redundant information is the delay introduced to the communication. In this case, the microcontroller registered a delay that never exceeds 35 ms. With regards to the jitter performance 12 ms for a non FEC connection and at 13.5 ms for a FEC connection are recorded. Although these values could influence a session's quality, they are compensated by a jitter buffer in the program.

A comparable result to data shown above is derived from the analysis of the received sequence number in real-time versus the number of received packets, as seen in Figure 3.11. As expected the greater the packet loss rate, the greater the slope of the response is. This is shown in the expanded figure of 15% of packet loss where a packet is lost between the sequences 90 and 100. Furthermore, TABLE 3.1 shows the recorded values for the experiments where the benefits of using FEC in percentages can be seen. In the packet loss column, the captured packet loss percentage in a real scenario matches the theoretical values corroborating that the system is running correctly. In the packet recovered column, values are almost halved in comparison to

Table 3.1: Results For A Wire to Wireless VoIP Session with FEC

Sent Packets	Lost (%) Packets	Packet Loss Recorded	Packet Loss Recorded (%)	Recovered with FEC Recorded	Recovered with FEC Recorded(%)	Efficiency (η)
1304	0	0	0.00	0	0	0.04
1028	5	62	6.03	30	2.91	0.75
1162	10	110	9.47	66	5.67	0.97
874	15	110	12.59	72	8.23	0.88
808	20	163	20.17	89	11.01	0.81

the packet loss rate value, leading to the conclusion that an extra $64/3 = 21.33$ kbps throughput required by FEC noticeably improves the system's overall performance. Therefore the FEC efficiency is defined from equation (3.3).

$$\eta = \frac{(\triangle MOS)}{(\triangle BW)}(MOS/Kbps) \tag{3.3}$$

The numerator ($\triangle MOS$) is the difference of FEC and non-FEC based MOS test normalized to the maximum value of MOS. The denominator ($\triangle BW$) is the bandwidth difference between FEC and non FEC system for a specific codec. In order to associate each codec's performance the numerator value is normalised to the bandwidth of the codec itself, i.e 64 in case of G.711. The equation represents how valuable is the extra information sent through FEC over the quality of voice received. Experimental results show that with 0% of packet loss the efficiency proves to be very low where theoretical response should be 0. However, the higher the packet loss the better the efficiency, in other words, the higher the losses the more worthy is to send redundant data. This efficiency parameter can be considered as a regulator for end nodes to evaluate the use of FEC under bursty losses.

3.5 Conclusion

In this chapter, an embedded SIP server Gateway to mobile networks with FEC is presented. Wired and wireless networks have been interfaced by developing an expansion board and integrating FEC system for VoIP sessions as an example of a heterogeneous network connection. Our development demonstrates that utilising FEC for a wired to wireless system, performs satisfactorily even under high packet loss channels. Furthermore, the efficiency cost of such model, as shown in TABLE 3.1, determines that under poor channel conditions the implementation of FEC is beneficial if action is taken by the gateway. Consequently it is proved that active SIP GWs can enhance end-to-end speech quality, however this is only approved for a single case scenario. Hence, an analytical model to predict speech quality perception under a number of different networks is required for the rapid assessment of QoS which is presented in the following chapter.

CHAPTER 4

Speech Quality Prediction for Heterogeneous Networks

4.1 Introduction

IN the previous chapter a VoIP GW implementation proved that passive GWs, i.e. only *store and forward* GWs, perform worse when compared to GW's with FEC that are aware of channel conditions. The results gained from such implementation highlight the advantage of intelligent networks to improve QoS. In next generation networks, end-to-end communications will utilise a variety heterogeneous technologies and consequently predicting quality performance can play an important role in network design. In Section 2.3, MOS and E-model showed that VoIP calls can be rated with subjective and objective methodologies respectively. In Chapter 3 the MOS model is used to present the results while in this chapter the E-model is considered to predict speech quality in heterogeneous networks. However, this analytical model presents a number of limitations where the restricted number of parametrised audio codecs , see recommendation [31], narrows down the scope of further research to be carried out. In addition, the model is based on a single audio codec connection from an end-to-end perspective which is acceptable for an Internet core soft-phone to soft-phone call but not for a heterogeneous communication system. Considering the implementation in Chapter 3 for a wired-to-wireless communication, where the wired network is provided with G.711 audio codec, packet losses rate clearly have an impact on overall performance. The crucial question is what happens if the wired network uses another audio codec? Is it necessary to set the implementation again or can speech quality perception be predicted? Current state of the art models do not provide a straight forward solution to this problem, however the analytical model presented in this chapter can predict speech quality perception under two independent channels using a self developed simulation tool.

In this chapter, the proposed model provides up to three audio codes and two cascaded network estimations at a time. Simulator response is tested with MOS and results are compared to the proposed E-model extension where the lower the error-margin the better the prediction. Channels are modelled through the Finite State Markov Chain (FSMC) analytical model due to extensive literary research that has proved it highly successful [72]. The analysis of the effect of multiple channels on speech quality is carried out in two parts. Firstly, single network tests are performed to compute codec specific parameters, i.e I_e and Bpl for G.711, GSM06.10 and Speex. Recall that the E-model uses I_e (equipment impairment factor) and Bpl (packet loss robustness factor) to parametrise an audio codec (see equation (2.12)). Secondly, double channel tests with codec and packet loss variations are tested and analysed following the implementation presented in the previous chapter. The results proved that low error-margins are achieved with the proposed method.

The remainder of the chapter is organised as follows. Section 4.2 describes the simulation system design, utilising audio codecs and the packet loss dependency over multiple channels for VoIP session. Section 4.3 presents the proposed formula to extend the E-model. Section 4.4 validates the proposed analytical model and evaluates the performance to predict speech quality under heterogeneous networks. Finally Section 4.5 concludes the chapter.

4.2 New Simulation Design for Multiple Channels

In this section, a new simulation design for multiple channels is presented. The idea is to design a generic system capable of modelling the behaviour of multiple different sub-networks that take part in an heterogeneous communication network as long as these sub-networks can be mathematically modelled. Figure 4.1 depicts the comparison between the conventional E-model and the proposed system in this chapter. The conventional E-model is based on a single network single audio codec prediction system. The model has been widely researched and its performance has been agreed to have a fair accuracy in the prediction of speech quality. The proposed model expands the conventional method to adopt a multiple number of networks and audio codecs where a Gateway provides the communication to the differing networks.

Consider the implementation in Chapter 3 where a VoIP connection from a soft phone to a mobile phone using a SIP proxy GW crossing IEEE 802.3 and the GSM networks is presented. Currently such connection can not be modelled using conventional E-model. To solve this problem audio codecs described in Section 2.2.3 on Page 36 are used with FSMC channel models to set a MOS test. Initially, the analytical model for multiple independent FSMC networks is shown and then the simulation tool is described.

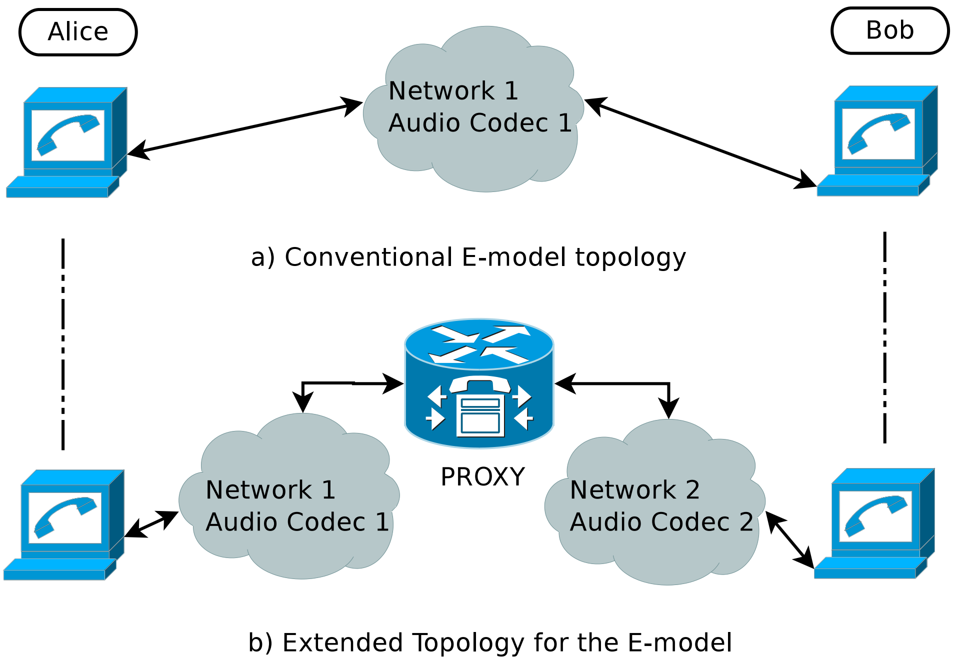


Figure 4.1: Comparison of the conventional E-model and extended E-model for multiple channels.

4.2.1 Markov Based Channel Modelling

In the previous chapter, the use of FSMC [62] has been presented as a model for bursty networks, where network's packet loss probability is related to the delivery or not of last packet. Recall from Figure 3.2 on page 66 that $P(G \rightarrow B) = p$ represents the probability of moving from the Good state to the Bad State whereas $P(B \rightarrow G) = q$ represents the probability of moving from the Bad to the Good state. Hence, the probability of being in either the *Good* or the *Bad* is calculated using $\pi_0 = (q/(p+q)), \pi_1 = (p/(p+q))$ respectively. Considering that the actual path of communication must travel across multiple channels, each of them can be represented by a first order FSMC as seen in Figure 4.2. This figure illustrates n number of two state Markov Chain model. The analytical expression of this communication system is defined by the transition matrix in equation (4.1) [73].

This matrix is a multiple independent FSMCs gathered in a single matrix where π_i is $P(G_i- > B_i)$ of the i^{th} channel and q_i $P(B_i- > G_i)$ for $(i = 1, 2, ...n)$. The average packet loss probability per channel is independent and therefore the overall average packet loss probability, Ppl_t, is calculated by equation (4.2) where ppl_i is

$$\begin{pmatrix} 1-p_1 & q_1 & 0 & 0 & 0 & 0 \\ p_1 & 1-q_1 & 0 & 0 & 0 & 0 \\ 0 & 0 & 1-p_2 & q_2 & 0 & 0 \\ 0 & 0 & p_2 & 1-q_2 & 0 & 0 \\ 0 & 0 & 0 & 0 & . & . \\ 0 & 0 & 0 & 0 & 1-p_n & q_n \\ 0 & 0 & 0 & 0 & p_n & 1-q_n \end{pmatrix} \begin{pmatrix} \pi_{1.0} \\ \pi_{1.1} \\ \pi_{2.0} \\ \pi_{2.1} \\ . \\ \pi_{n.0} \\ \pi_{n.1} \end{pmatrix} = \begin{pmatrix} \pi_{1.0} \\ \pi_{1.1} \\ \pi_{2.0} \\ \pi_{2.1} \\ . \\ \pi_{n.0} \\ \pi_{n.1} \end{pmatrix} \tag{4.1}$$

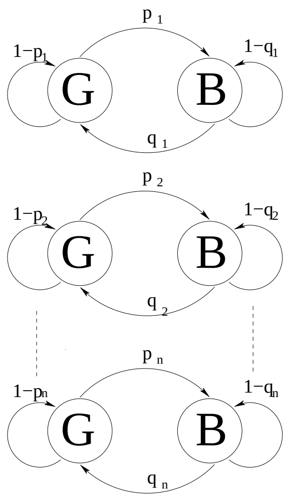


Figure 4.2: N number of independent FSMC channels.

$FSMC_i$'s average packet loss probability which depends on $\pi_{i.0}$ taken from equation (4.1) defined from equation (3.1) [73].

$$Ppl_t = 1 - \prod(1 - ppl_i) \tag{4.2}$$

4.2.2 Simulation Tool

Figure 4.3 on the next page shows the developed simulation tool for speech prediction tests, written in C language. Voice is sampled to 8 KHz, 16 bit, Little Endian, grouped in 160 samples for a 20ms packet size. Packets are encoded, filtered by Markov Chain (MC_i) model and buffered at the receiver side to avoid jitter distortion. A decoder is applied (if necessary) and encoded again to be filtered by the next channel model. Note that the process to exchange different codec types is done by an entity called

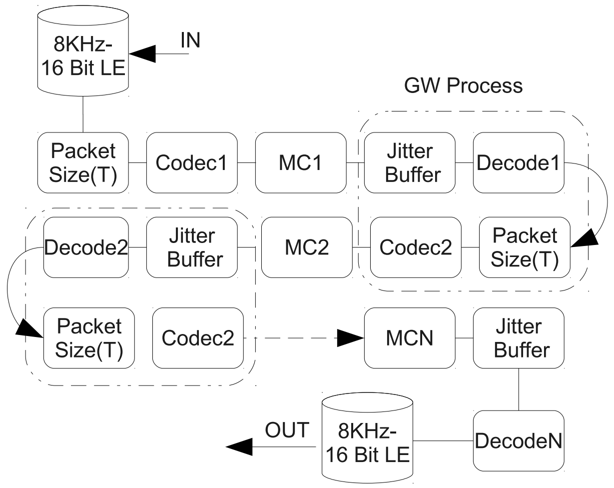


Figure 4.3: VoIP session model for Multiple channels [73].

Gateway (GW). When the packets arrive at the final destination they are decoded and played back as 16 bit, 8KHz sampled signal. The model is flexible to be set from 1 to n number of channels, although in these experiments only up to two channels have been investigated.

4.3 Proposed Extended E-model

The new simulation design in the previous section, presents an scenario where n number of channels with n number of audio codecs is possible. This new paradigm introduces a number of constraints to predict speech quality where conventional E-model described in [2] is not valuable any more. In this section, the conventional E-model is extended based on extended trial and error approaches that minimises the difference between subjective experiments and the mathematical model itself. The proposed new *R-value* is defined as follows [73]:

$$R = R_0 - I_S - I_D - I_{e-eff_t} + I_{eq} + A \tag{4.3}$$

where the signal to noise ratio, R_0, the simultaneous impairment factors, I_S, and the advantage factor , A, are evaluated using default values as shown in equation (2.5). Equally, the delay impairment factor, I_D, is also defined according to the conventional method (see equation (2.6)). This is conceivable because this impairment is independent of channel status or utilised audio codec type. Hence, the extended *R-value* can be rewritten as [73]

$$R = 93.3 - I_D - I_{e-eff_t} + I_{eq} \tag{4.4}$$

The proposed equation differs in a number of aspects from the conventional E-model. The effective equipment impairment factor, I_{e-eff_t}, is defined as the number of factors that predicts the impairment factor related to channel response with different audio codecs. Recall that I_{e-eff} uses Ppl and $BurstR$ to model the channel and I_e and Bpl for audio codec performance (see equation (2.12)). In a communication system with multiple channels and multiple audio codecs, the overall performance is always dependant on the poorest audio codec and channel. In a parallelism with a production line, the poorest audio codec is presumably the bottleneck of the system whereby productions maximum output is the output of the bottle neck production. Consequently, the proposed analytical model for I_{e-eff_t} is based on the poorest I_{e-eff_i} within the communication system to predict over all performance where the poorest audio codec is determined from the I_e values [73].

$$I_{e-eff_t} = I_{e-eff_1} + \sum_{i=1}^{n} \rho_i \times I_{e-eff_{i+1}} \tag{4.5}$$

where

$$\rho_i = \begin{cases} 1 & \text{for } I_{e_i} = Ie_1 \\ \frac{I_{e_i}}{I_{e_{i+1}}} \times 100 & \text{for } I_{e_i} \neq Ie_{i+1} \end{cases} \tag{4.6}$$

and

$$I_{e_1} > I_{e_2} > ...I_{e_n} \tag{4.7}$$

$$BurstR_i = \frac{p_i}{p_i + q_i} \tag{4.8}$$

In this case, I_{e-eff_1} refers to the poorest audio codec out of n number of channels and ρ is a factor that allows diminishing the impact of other codecs degradation in comparison with the performance of I_{e_1}. It is important to note that multiple channels with the same audio codec, i.e. $I_{e_i} = I_{e_1}$, are predicted as a sum of independent effective equipment factors. The burstiness modelled with FSMC is represented by equation 4.8 where p_i refers to the average packet loss percentage of the channel and q_i is the parameter that defines the channel itself.

In addition to the extended effective equipment factor, the proposed analytical model adds a new impairment factor, defined as the distortion equaliser impairment factor, I_{eq}. This factor has been incorporated to the analytical model due to inaccuracies obtained with very high packet losses, i.e. packet losses above 15%. The main reason behind this parameter is that human perception barely distinguishes between a communication system with 20 % and 15 % of packet looses resulting in large difference between MOS tests and proposed analytical model. The new impairment factor is defined as [73]:

$$I_{eq} = 1 + e^{\frac{Ppl_t - 15}{15}} \tag{4.9}$$

where Ppl_t follows equation (4.2) and 15 refers to a fixed numerical value defined after substantial experimental results.

4.3.1 Methodology to determine audio codecs' parameters

The above formula extends the conventional E-model prediction to heterogeneous networks which is tested in the following section. Nevertheless, the audio codecs described in Section 2.2.3 are not defined by the ITU-T recommendation [31] and hence experimental measurements are required to determine their parameters.

An audio codec in the E-model is defined by I_e and Bpl. To determine the value of I_e, MOS tests for each audio codec are carried out with zero delay and zero packet loss. The experimental values are converted to the *R-value* which is set with default values. Since $Ppl = 0$ and the rest of the parameters are known, I_e is calculated by using the following equation:

$$I_e = 93.3 - R - I_D \tag{4.10}$$

Once I_e is defined, the value of Bpl is determined. In this case, MOS tests are performed for each audio codec with a variety of packet loss percentages using a MC model with $q = 0.7$. The reason behind this q value is that previous chapter's implementation is based on this parameter which provides results with a closer match to real implementations. The packet loss range is varied from 5% to 20% where each time the Bpl value is measured as

$$Bpl = \frac{(95 - I_e)Ppl}{93.3 - R - I_D - I_e} - \frac{Ppl}{BurstR} \tag{4.11}$$

Obtained values from these results are averaged and presented in the following section.

4.4 Results

The simulation tool in Figure 4.3 is used to record experiments with different audio codecs and packet loss models. Recorded files are uploaded to our website [74] and MOS tests are carried out to assess the speech quality. Results are converted to the *R-value* with equation (2.3) to compare the results. Using the extended E-model and parameters that correspond to each recorded file, the differences between the experimental results and analytical values are presented in this section. MOS tests are carried out for over 20 people from different gender, age and nationality. The tests' statistical values are shown in Table 4.1 where parameters are defined in rows and the columns correspond to the performed tests as explained hereafter.

The results are divided in three subsections. The first part is based on a single network performance using the conventional E-model method to determine audio codec parameters. The second part is based on experiments with two channels which is also known as two hop connection. The results include two options: on one hand two channels with different packet loss rates considering that both networks agreed using the same audio codec and in the other hand two channels with different packet loss rates and audio codecs.

Table 4.1: Statistical values for the MOS test, where Diff. refers to Different and values are presented in percentages (%) [73].

Paramaters	Options	Single Network	2 Networks Equal Codec	2 Networks Diff. Codec
Gender	Male	61.23	77.05	62.36
	Female	38.77	22.95	37.64
Age	0 to 20	0	0	0
	21 to 30	90.88	91.80	100
	31 to 40	9.12	8.20	0
	41 and above	0	0	0

4.4.1 Validating the Model

The proposed model for heterogeneous networks is validated with the hardware implementation presented in Chapter 3 as shown in Figure 4.4. On one hand, the results that refer to *G.711-GSM 0610 Hardware* are taken from Figure 3.10 on page 78 using the data with regards to *G711 to GSM with no-FEC*. These values have been translated to the E-model using equation 2.3. On the other hand, the results obtained by the proposed model for cascaded network performance using the mathematical model described in previous section, are represented under the name of *G.711-GSM06.10 Model*. Comparing the results of both experiments, it is concluded that the model is a good and fair approach of the hardware model since the differences between both results is negligible. Hence the variety of the results obtained throughout this chapter are considered as realistic as hardware implementation results for speech quality prediction.

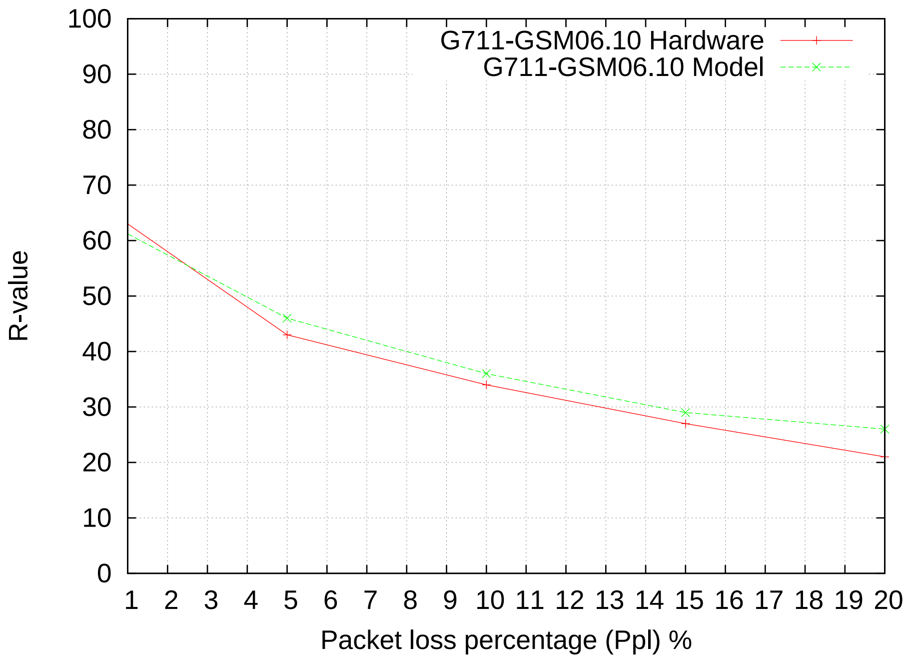


Figure 4.4: Comparison of Hardware vs Model.

4.4.2 New Parameters for Audio Codecs

In these tests a single network with single audio codec is tested in order to define the parameters for G.711, GSM and Speex audio codecs. The simulation tool is set to $q = 0.7$ [53] and no delay is introduced. The obtained results are presented in Figure 4.5 and Table 4.2 following the formulae in Section 4.3.1. On one hand, the figure illustrates the average performance of each codec whereas the table presents I_e and Bpl parameters for each audio codec which are divided in two columns. First column represents the already available parameters from ITU-T recommendation [31] where unknown values are shown with '-' symbol. Second column is the experimental results obtained using equations (4.10) and (4.11). Note that the I_e difference for G.711 between the experimental results and values from [31] is negligible whereas the Bpl shows higher differences. This is due to the fact that measurements are carried out with FSMC models rather than random losses. Regarding to GSM audio codec it can be seen that the I_e differs slightly from the values provided by the ITU-T, which ensures that the experimental results and the methodology to obtain the values are accurate. Overall, robustness values appear to have high value for linear predictive codes (Speex and GSM06.10) however, the codec performance adjusts to the obtained

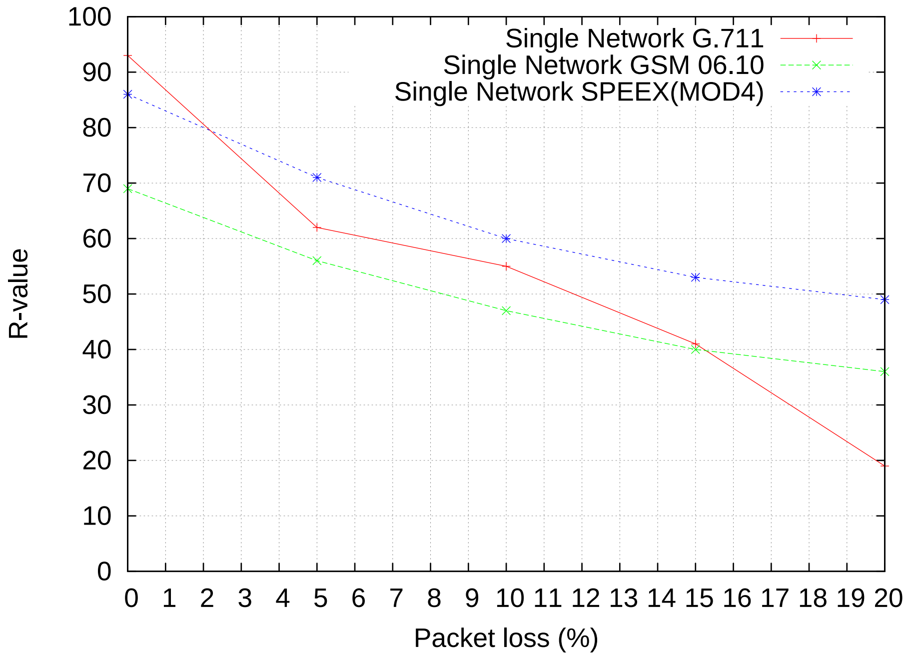


Figure 4.5: Single Network codec performance results [73].

measurements as seen in the following subsections.

Table 4.2: Codec Measurement Results [73].

Reference	ITU-T [31]		Experimental	
	I_e	Bpl	I_e	Bpl
GSM06.10	20	-	24.37	39.32
Speex-MOD4	-	-	5.37	37.82
G.711	0	4.3	0.37	12.92

4.4.3 Speech Quality Prediction for a Two Hop Connection

In this set of experiments, two channels are tested with MOS, converted to the *R-value* and compared to our proposed model. The results are based on two independent FSMC models with two variations: one for equal codecs per channel and the other for different codecs.

4.4.3.1 Equal Audio Codecs per Hop

Equal codecs per hop experiment is based on two FSMC models where q_i value is fixed to 0.7 in both channels and two different packet losses per channel, ppl_1 and ppl_2, are configured at the simulation tool. Ppl_t is calculated using equation (4.2) with $i = 2$ whereas $Ie - eff_t$ is calculated with equation (4.5) where $\rho = 1$ since both channels' I_e is the same. Finally, audio codec parameters are taken from Table 4.2.

Figure 4.6 shows the performance graphs for different packet loss channels and audio codecs. Each graph is represented by a fixed packet loss rate depicted in the upper left label and a variable packet loss rate in the x axis. In the y axis the *R-value* of the analytical model versus the experimental results is illustrated. All graphs include the three audio codecs: G.711, GSM and Speex, where *audio˙codec˙name E-model* refers to the proposed extended model and *audio˙codec˙name MOS* is the experimental result obtained through the website.

G.711 performance with an error-margin average of $5.02R$ and a maximum error of $9.54R$ shows the consistency of the proposed formula. The maximum error appears at $Ppl1 = 20\%$ & $Ppl2 = 5\%$ where a rather exceptional case occurs. In this error percentage, the tested file suffers from a combination of packet losses where a number

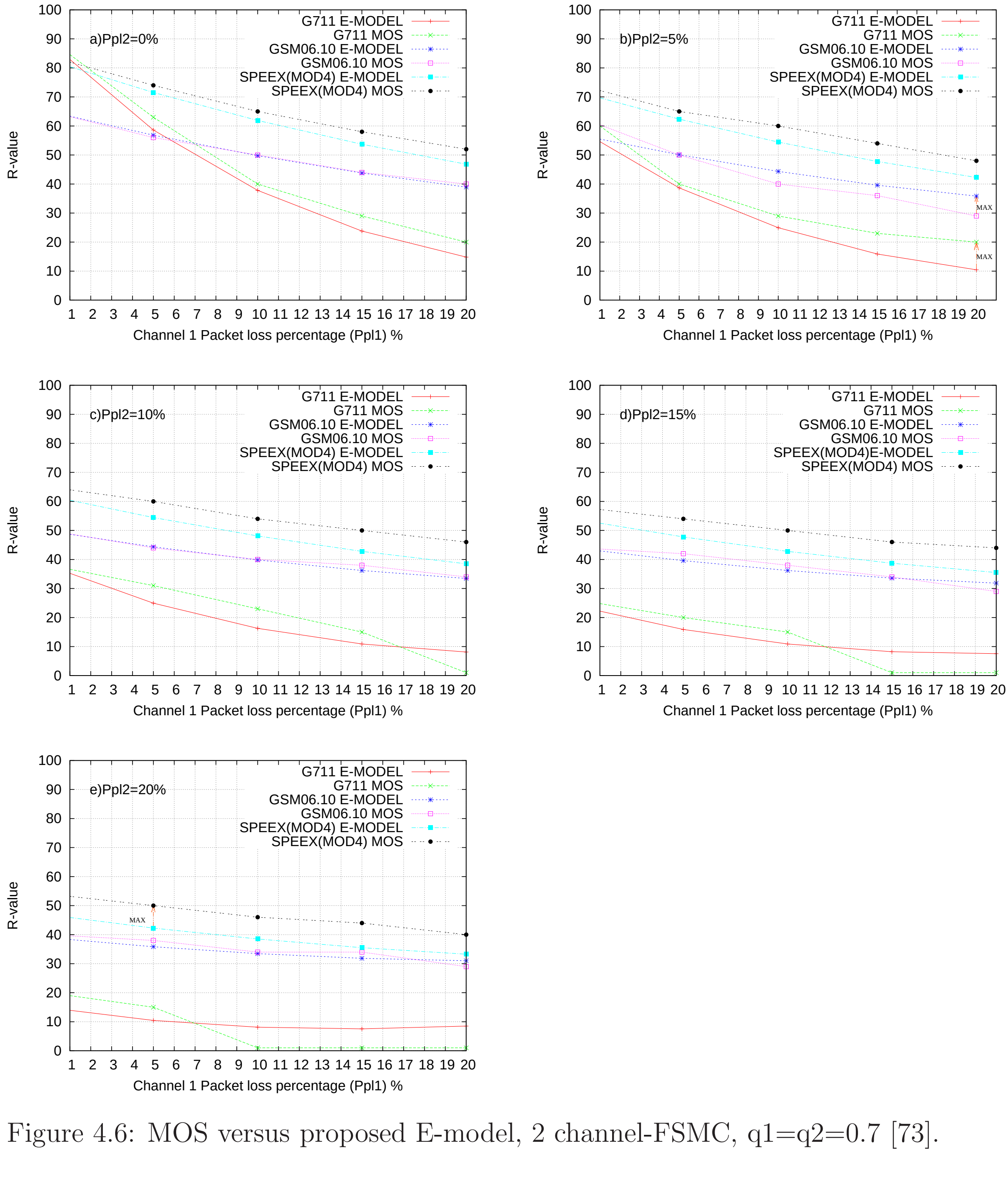


Figure 4.6: MOS versus proposed E-model, 2 channel-FSMC, q1=q2=0.7 [73].

of short words (1 syllable) are lost and the comprehension of the speech becomes hard, resulting in a lower rate in the MOS test. Conversely the mathematical models do not distinguish between words or packets and therefore the estimated speech quality is slightly better than that of users perception. As the packet percentages increases, MOS tests show that users barely distinguish tests with 15% of packet loss rates or higher rates. This can be seen with $Ppl1 = 15\%$ & $Ppl2 = 15\%$ and $Ppl1 = 10\%$ & $Ppl2 = 20\%$ where the MOS is dropped to one which is equivalent to the lowest value of the MOS system.

GSM 06.10 codec error-margin average is $1.68R$ the lowest of all of them. The smooth response obtains the largest error in $Ppl1 = 20\%$ & $Ppl2 = 5\%$ where the value rises to $6.84R$. Again this is related to the same problem described for G.711, where the loss of a number of consequent words generates a bad performance for the users. Despite of it all, the performance of the analytical model is very accurate in this case.

Finally, Speex performance shows an average error-margin value of $3.03R$ with a maximum peak of $4.92R$ recorded at $Ppl1 = 5\%$ & $Ppl2 = 20\%$. In this case, the error average of the analytical model is always marginally more positive than the actual user perception but due to low error differences this is considered insignificant.

These experiments demonstrate that in the scenarios where both channels have the same audio compressors, predictions are accurate compared to those values obtained through subjective testing.

4.4.3.2 Different Codecs per Hop

Different codecs per hop experiment is based on two FSMC models where q_1 and q_2 values are set to 0.7 and 0.1492, respectively. The former value represents an Internet connection [53] whereas the later is taken from research based on GSM networks [75]. Different packet losses and audio codecs per channel are utilised as input sources of the simulation tool. Ppl_t is calculated using equation (4.2) with $i = 2$ whereas $Ie - eff_t$ is calculated with equation (4.5) where ρ value depends upon the audio codecs considered per channel.

Figure 4.7 shows the performance graphs for different packet loss channels and audio codecs. The representation of each graph follows the equal codecs graphs shown in the previous experiment. In this case, experiments for G.711 with GSM, G.711 with Speex and Speex with GSM are presented. It is important to point out that the order of the audio codecs is considered irrelevant since the analytical model takes into consideration the poorest audio codec to predict speech quality.

G.711 with GSM error-margin shows an average of $8.35R$ and a maximum error peak of $22R$. This combination of codecs illustrate the poorest performance all results for the proposed analytical model where the peak can be found at $ppl_1 = 15\%$ & $ppl_2 = 20\%$. There are two main reasons related to this discrepancy: firstly from previous experiments is understood that human perception for very high packet losses tends to be distorted due to the incapacity of humans to distinguish between bad and very bad quality. Secondly, it is believed that the combination of LP codecs with waveform codecs is expected to perform poorly due to the difference in the I_e parameter which is reflected in the lower values obtained by the mathematical model.

The combination of G.711 and Speex presents a performance of $2.28R$ average error-margin with a maximum peak of $8.05R$. The peak is found at $ppl_1 = 20\%$ & $ppl_2 = 0\%$ which if compared with the performance of G.711 with GSM it can be seen that the high percentage of packet losses in G.711 has an impact on LP based audio codecs. Nevertheless, the performance of the analytical model can be considered as fair due to low average error-margin.

Finally Speex with GSM shows a performance of $3.71R$ error-margin error with a peak of $8.55R$ at $ppl_1 = 15\%$ & $ppl_2 = 0\%$. In this case, the lower quality obtained through Speex codec affects the natural performance of GSM and therefore the MOS test shows lower values than the analytical model. Despite of it, the average error-margin is low and proves the proposed analytical model to be a good approach.

Considering all the results, several conclusions that can be drawn. Initially, wavelength based speech codecs always have the best audio performance. This is observed in both cases where $ppl_1 = 0\%$ & $ppl_2 = 0\%$ where results with G.711 codec show the best *R-value*. The second conclusion is that very high packet loss rates are not distinguishable by human perception regardless of the audio codec utilised. This is interesting because intermediate nodes could take action by forcing to either end a VoIP session or consider FEC systems if the expected overall quality drops down to 40R or below. The third conclusion refers to cases with different audio codecs are utilised where two LP based audio codecs perform worse than a wavelength and a LP based calls. In fact, throughout all these experiments the best performance is given to G.711 with Speex codec. Finally, although few examples of high error-margins are found, it is expected that the on going test results will corroborate even greater the

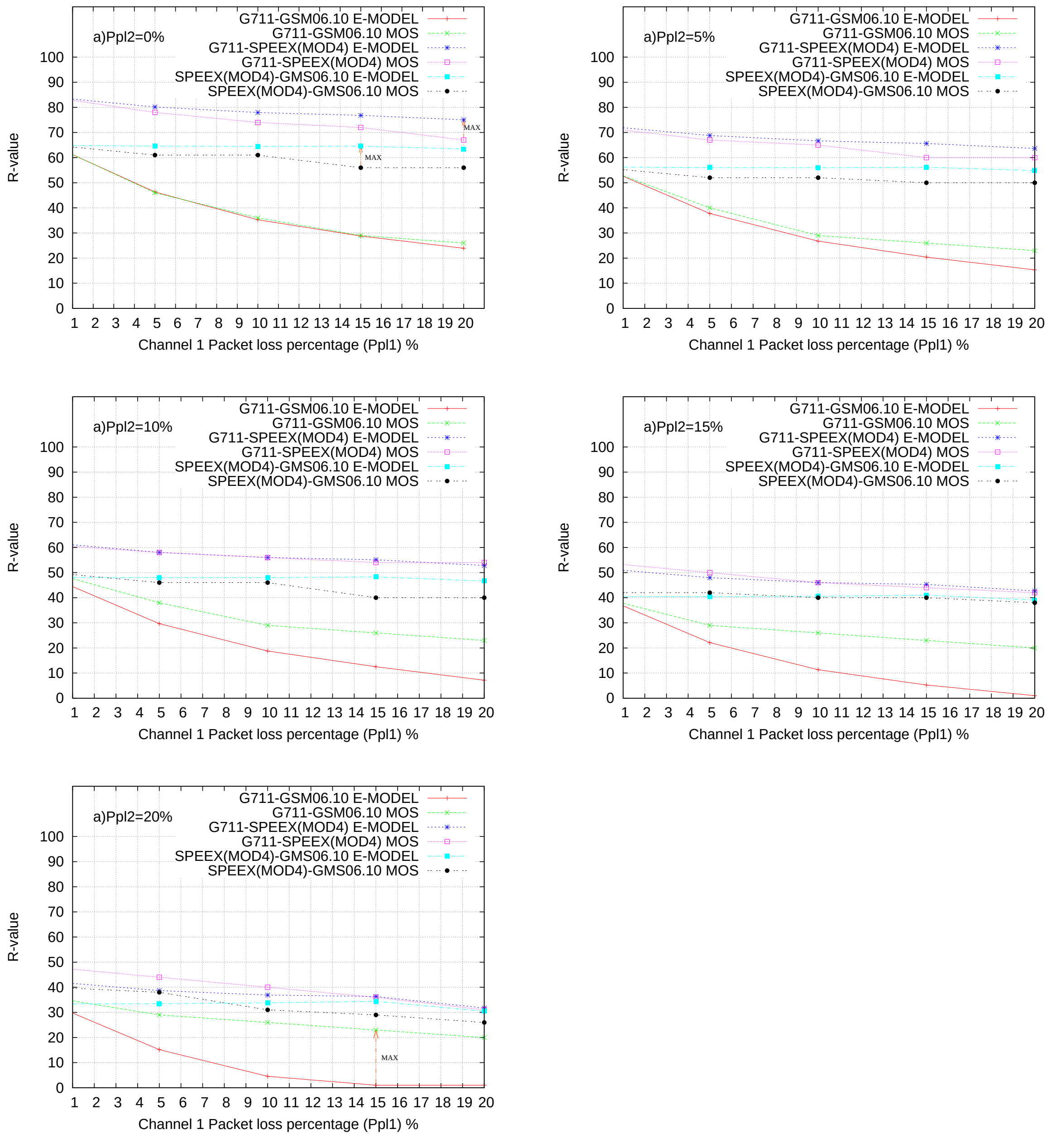


Figure 4.7: MOS versus proposed E-model, 2 channel-FSMC, q1=0.7, q2=0.1492 [73].

formula presented in this thesis.

4.5 Conclusion

In this chapter, an analytical study for VoIP with two network channels is presented based on the current E-model. Firstly, the study presented single network results where three audio codecs are tested. I_e and Bpl values for them are measured using FSMC models and validated with the current G.711 codecs parameters as defined by the ITU-T. Secondly, two channel network experiments are carried out with two major set-ups:

1. Equal audio codecs in both channels

2. Different audio codecs per channel

The proposed extended E-model is used to match the MOS test by introducing two main variables: a new impairment factor to equalise results with high packet loss rates, and the ρ factor which balances the performance of independent channels with different audio codecs. The average error-margin of the analytical model compared with the MOS values shows a reasonable 1.68R to 8.35R error-range.

CHAPTER 5

Network Coding on VoIP

5.1 Introduction

IN telecommunications there are three types of communication methods: unicast-to-unicast, unicast-to-multicast and multicast-to-multicast. The telephone network is based on a unicast-to-unicast method belonging to a hierarchical topology whereas a broadcast network, based in a flat topology, allows communications between all peers facilitating unicast-to-multicast and multicast-to-multicast methods. Nowadays, broadcast networks are preferred, specially wireless broadcast networks because they cut down costs to service providers for the last mile connection.

Wireless networks often function on omnidirectional basis however, rather than making use of such beneficial characteristics, most technology has been designed to achieve unicast-to-unicast communications. This is understandable when taking into consideration the security issues relating to private or personal information and its protection against fraudulent use. This can be seen in home networks where the core of the Internet remains mainly unicast communication. Conversely services such as satellite TV have already developed a unicast-to-multicast service.

Network coding addresses unicast-to-multicast and multicast-to-multicast communications from an unknown angle, as seen in Section 2.4. Initially Yeung and Zhang presented a pioneering research on unicast-to-multicast for satellite communications in [35]. This concept was further developed by Ahlswede et al. in [32] in a later paper naming the method as "network coding". This inspiring method makes use of coding theory to increase channel throughput by combining existing packets in the network and multicast them so that receivers can decode the message. This theory is particularly encouraging for an increase in performance of half duplex networks and it is not restricted to any particular layer of the Open System Interconnection (OSI) model.

VoIP suffers from long delays caused by queues that converge into packet losses resulting in poor call performance. To insert network coding schemes and create even more delays at first glance might appear counterproductive but in this chapter, a new proposal to use network coding theory for VoIP services is presented without dependency on the physical layer. The block diagram of a possible case scenario where the proposed model can be implemented is shown in Figure 5.1. In this illustration a base station with multiple cells is connected to a network with a SIP proxy and a Media Gateway that provides VoIP services. The users are connected to the base

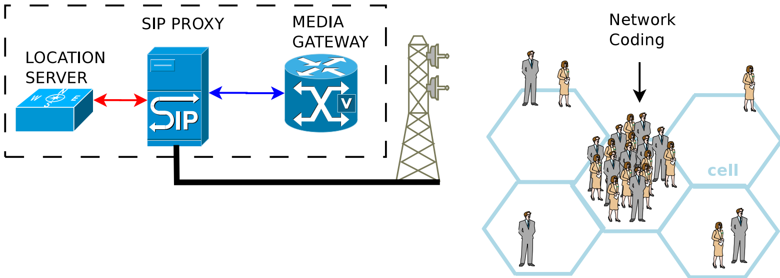


Figure 5.1: Scenario where Network Coding for VoIP is an improvement.

station which divides the geographical area into cells. In particular, one of the cells, registers a large number of user as it could be the case when popular events happen in a concentrated area, for example, a football match or a music concert. Subsequently the aim of this research is to improve the performance of conventional VoIP services with special care at last mile wireless networks. The theoretical overview described in Section 2.4, presents a broadcast network graph model using acyclic networks. One such network, as shown in Figure 2.13 on page 57, was considered whilst designing a new paradigm for VoIP communications to support network coding techniques. This paradigm requires a set of changes: a packet modification, queue systems and call flow synchronisation. This new algorithm for VoIP, remarkably increases by 100 % the number of VoIP calls that a Basic Service Set (BSS) can handle.

The remainder of this chapter is organised as follows. Section 5.2 presents the proposed algorithm for VoIP with network coding. Section 5.3 defines theoretical minimum delays for VoIP with QoS followed by an applied analysis to IEEE 802.11b networks. Section 5.4 implementation and 5.5 its results, describe and discuss the theoretical models and obtained performance. Finally, Section 5.6 concludes the chapter.

5.2 An algorithm for VoIP with Network Coding

In this section, a new proposal for VoIP services is presented, fully compliant with current state of the art and independent of the physical layer.

VoIP requires a large scale of network resources to both Internet Core and end users. The traditional method for VoIP is based on central nodes called SIP Proxies with user Registrar and Location Service (LS) capabilities. End users make use of

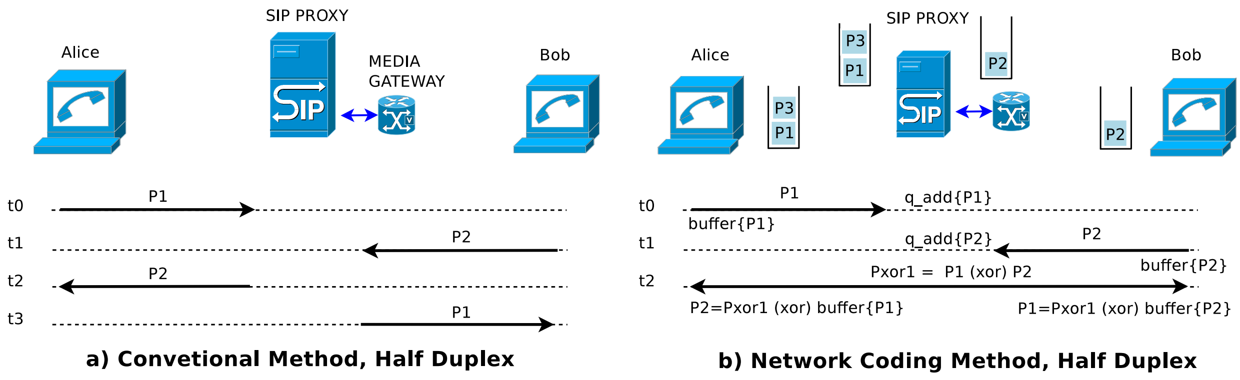


Figure 5.2: Design comparison between conventional method and our proposed method over a half duplex channel [76].

SIP Proxies to locate the call receiver; depending on session features, calls would be routed either from end user to end user (Peer to Peer system as seen Figure 2.2 on page 33) or through a centralised server as seen in Figure 2.3 on page 34. The former releases the traffic load to the central server and the latter keeps control of all data exchanged between users. In some cases, such as wireless communications, end users are connected to their base station and communication between end nodes must go through a centralised node. Conventional base stations will simply forward the message, missing the opportunity to maximise throughput of the channel applying network coding techniques.

Consider a SIP Proxy that acts as information relay between two end nodes. The conventional method would only forward packets from node A to node B by simply repeating them. With network coding central node can combine received packets to create a new set of outgoing packets reducing the usage of the network as explained in Section 2.4 on Page 50. In this case, we use the model of Figure 2.13 on page 57 to represent the real environment depicted by Figure 5.2 [76].

In this figure, *Alice* and *Bob* are defined as the sink nodes from Figure 2.13 on page 57 and the SIP server is the central node that acts on behalf of the source node and node 1. The information units are set with the simplest field, $\mathbb{F}_2 = \{0, 1\}$, the matrix encoders $M_{s,j}$ and $M_{1,l}$ are used as defined in equation (2.21) and (2.25), respectively and hence the broadcasted information is a modulo two addition in accordance with equation (2.29).

Illustration on Figure 5.2, compares the packet flow for a conventional centralised VoIP session against network coding method. The former implies that all packets from *Alice* to *Bob* and *Bob* to *Alice* are sent to the server and then forwarded according to its routing table values. This method would be suitable for a Gateway which interconnects networks handled by different signalling systems, for example the implementation in Chapter 3. Conversely, it would be an inefficient approach if the network between *Alice* and *Bob* is a broadcast network, such as Wi-Fi, WiMax or Ethernet. In these cases, central nodes could perform a linear encoding between received packets providing that end users keep a record of their own packets, represented by *buffer*{*P1*} and *buffer*{*P2*} for *Alice* and *Bob* respectively in Figure 5.2 (b). As shown by the illustration the impact on throughput and time sequence of network coding can be considerable if compared to a half duplex conventional method at a specific downlink. On the other hand, if full duplex channels are in use, the time sequences for network coding and conventional method are the same, as it is shown in Section 5.5. In spite of this, the traffic load with network coding remains more efficient in terms of throughput than conventional methods.

The encoded new packet is delivered to end users by multicasting it, which implies that end nodes must be aware of which IP and port they should join to. This is achieved by using SDP values where network coding support is specified; by default the destination port is chosen as the session initialiser port, i.e. in case of Figure 5.2 (b) *Alice's* source port. Although multicasting has been mainly dedicated to voice and video on demand, in this case, since the packet is encoded, only both end users are able to decode it, inheriting an extra layer of security to prevent eavesdropping and man in the middle attacks.

5.2.1 Packet Flow

The flow chart in Figure 5.3 represents a phone call from *Alice* to *Bob*, assuming that both end users have been previously registered against the SIP proxy. *Alice* generates a phone call to *Bob* by sending an *Invite* request including the options for audio codecs and network coding support (*SDP:NTWCDN*). The SIP server checks the registered users table to ensure that the request belongs to an internal user as well as ensuring network coding compatibility. If any of these two conditions fails the phone call will be rejected with the appropriate message response, for instance *606 Not Acceptable*. The server sends a *100 Trying* response to advise *Alice* that the call has been processed. *Bob*'s UA is active and therefore sends a *180 Ringing* message

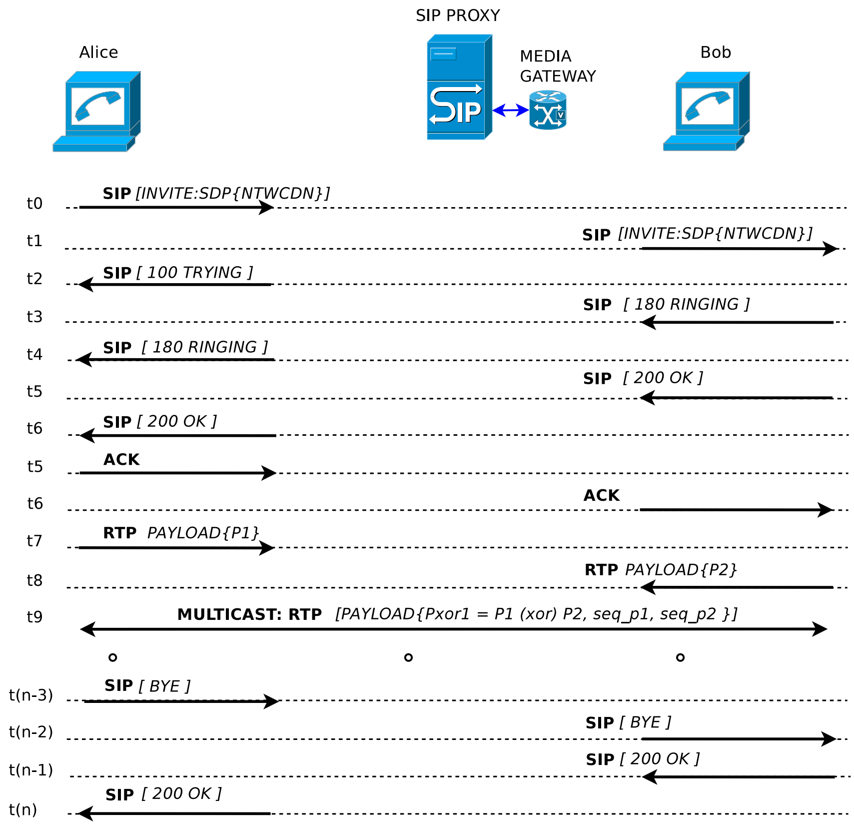


Figure 5.3: Packet flow of multicasting VoIP with network coding schemes [77].

followed by a call acceptance *200 OK* message. In this last message, *SDP:NTWCDN* is appended and monitored by the proxy in case it is not compatible with SIP proxy's configuration. Once the signalling system has established the parameters, *Alice* and *Bob* can start talking. On one hand the UAs, *Alice* and *Bob*, send messages to the central node keeping an internal buffer with sent packets. On the other hand, central node computes a bitwise *xor* with received packets, sending the outcome by multicast to both end users. Both end users will then decode the message *xor*-ring the received packet and their internal buffer packet. The conversation ends, in this example, by *Alice* sending a *Bye* request. This request is forwarded to *Bob* which replies with a *200 OK*, terminating the conversation.

5.2.2 Packet Layout

In a conventional system (see Figure 5.2 (a)) where *Alice* and *Bob* send $P(Alice)_i$ and $P(Bob)_i$ number of packets, for $i = 1, 2, 3...n$ respectively, the network would handle

$2(P(Alice)_n + P(Bob)_n)$ *Packets per conversation (Ppc)*. Since the number of packets on a constant bit rate is equal for both users, i.e. $P(Alice)_n = P(Bob)_n = P_n$, then $Ppc = 4 \cdot P_n$ where P_n is the number of packets sent from a single user. In this chapter, UAs uplink is considered as regular VoIP session that sends packets to the central server, thus *Alice* and *Bob* are sending packets as they would in the conventional format to their servers. The SIP Proxy uses the messages obtained from *Alice* and *Bob* to generate a new information packet $P(xor)_i = P(Alice)_i \oplus P(Bob)_i$ sent to end user by multicast. Network load is then given by equation (5.1) where $P(xor)_n$ is the number of packets sent from the proxy to the UAs. This is in accordance with the theory described in Section 2.4 on Page 50 whereby the maximum flow of the network is achieved by the new proposal [78].

$$Ppc = P(Alice)_n + P(Bob)_n + P(xor)_n = 3 \cdot P_n \tag{5.1}$$

In order to retrieve $P(xor)_i$ packets, *Alice* needs to know which $P(Alice)_i$ was used by the server. This is important as packets sent by the UA might be lost, forcing the SIP Proxy to use a previous packet for the encoder. Figure 5.4 illustrates the modified packet format to be applied to downlink packets; The RTP header follows the recommendation in [16] and adds, by padding, two numbers at the end of the packet: source sequence number and destination sequence number. These sequence numbers are the packets sent from the UAs that were captured and used for the encoder, i.e $P(Alice)_i$ for *Alice's* case. The node that initiates the session (sending an *Invite* request to SIP Proxy) is considered to be the source node whereas the receiver is the destination. Finally, following the recommendation from RFC 3551 [16] a last padding octet is added to specify the number of octets added to the standard payload size. This design allows the end node to synchronise received values with those stored at a local buffer.

Multicasting *xor*-ed packets adds an extra delay at the proxy server but does not overshadow its multiple advantages. The method is independent of the link layer and therefore easily applicable to any environment where a broadcast network is the medium. From previous research of VoIP capacity in broadcast network and primarily in wireless networks, down link throughput and delay is the bottle neck of the system [79]. Although extra 5 bytes at the header level is introduced to achieve synchronisation, throughput maximisation is guaranteed reaching a reduced load of %50 for a full VoIP session in the downlink. Furthermore the proposed design

```
Bytes    0                  1               2                   3
Bits      0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
         +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
         |V=2|P|X|  CC   |M|     PT       |         sequence number       |
         +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
         |                            timestamp                           |
         +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
         |           synchronization source (SSRC) identifier             |
         +=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
         |            contributing source (CSRC) identifiers     ....     |
         +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
         |                            AUDIO DATA                          |
         +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
         |       source sequence        |         destination sequence    |
         +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
        |   Padding        |
         +++++++++
```

Figure 5.4: Our proposed packet format for RTP [76]. V = version, P = Padding, X = extension, CC = CSRC Count , M = Marker, PT = Payload Type.

is fully compatible with existing RTP and SIP/SDP packet headers and could be implemented with no major changes.

Although throughput performance is a clear advantage when network coding is in use, delay introduced by the network propagation and encoders must be considered. In the next section minimum delay analysis is applied to IEEE 802.11b networks.

5.3 Theoretical Minimum Delay

In this section, we analyse the theoretical Minimum Delay (MD) of the proposed design and compare it with the MD of the conventional method for a Constant Bit Rate (CBR) VoIP session. Since MD determines the performance of a single VoIP session, the number of calls that the network can handle can be derived from it, i.e. the capacity of the network.

To derive the Delay Lower Limit (DLL) for a VoIP system with network coding it is necessary to define the achievable MD. This is only possible if the system is in the best case scenario, i.e. the channel is an ideal channel with no errors and at each transmission cycle there is only one active station to transmit a packet while other stations can only receive packets and provide acknowledgements if required by the link layer.

In a CBR system VoIP stations generate a packet every packetization interval. For an infinite long phone call if this packet is not transmitted before a new packet is generated the queue at the source station will be infinite too. Since this is not realistic it is assumed that transmission delay must be lower than the transmission time [79].

$$T_t < T_g \tag{5.2}$$

where T_g stands for packetization delay and T_t for total transmission delay of one voice packet per user including all overhead. Considering that our case scenario is a star topology network where an Access Point (AP) is the central node and users are within the service set (see Figure 5.2), then each VoIP packet has to be transmitted from one station to the AP firstly and from the AP to the next station secondly. On one hand, on a half duplex network this means that the network is busy four times for a packet exchange between users and therefore $T_t = 4 \cdot T_p$ where T_p is the delay introduced by a single standard packet. On the other hand, a full duplex network requires $T_t = 3 \cdot T_p$ because packet exchange between pair of network users can be done simultaneously and bidirectionally as described in Section 5.5. Network coding method implies that two packets are sent through the network as it would be the case in a conventional system. There is an additional delay of waiting time until both packets are received and a response message with the modulo two addition of both messages, including the extra overhead i.e. 5 bytes in our proposal. Equation (5.2) for network coding can be rewritten as [78]:

$$T_g > (\tau \cdot T_p) + T_{ntw} \tag{5.3}$$

where τ is the number of independent packets required to perform network coding, i.e. $\tau = 2$ in our case, but it could be larger if other coding methods are used at the central node and T_{ntw} refers to network coding delay introduced by the waiting time of two arrival packets and the encoded message response with extra overhead. Figure 5.5 illustrates a generic packet transmission for a network coding system. Packets are received in two different queues but from the same broadcast network which means that packets to these queues can not arrive at the same time. Once a packet is received modulo two addition is performed and the new packet with synchronisation overhead is sent as a multicast to all stations. The illustration shows that the transmission delay of a conventional voice packet, received at each input queue, consists of a MAC layer overhead T_{oh} and voice packet delay T_{data}. T_{oh} depends on the physical and link layers whereas voice packet delay is considered as the addition of overheads from the network

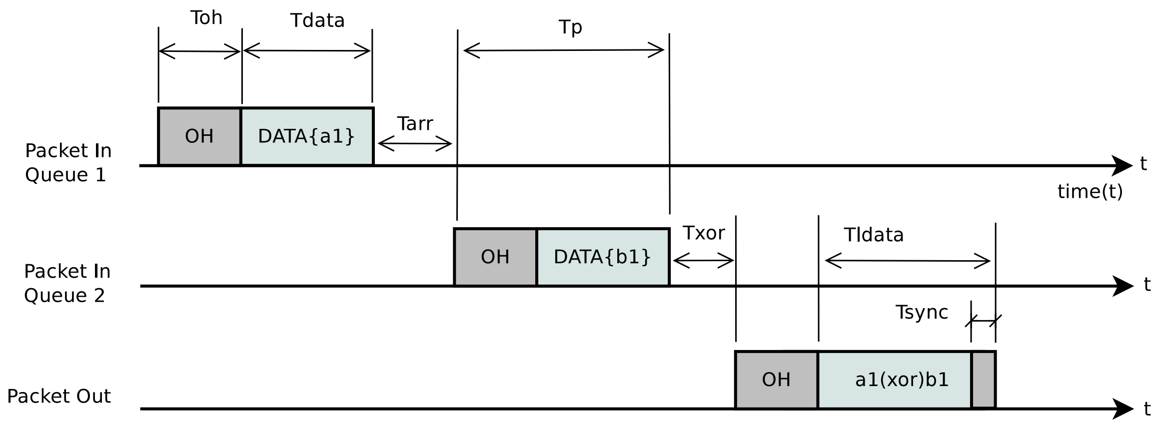


Figure 5.5: Transmission of a packet with network coding [78].

layer and above i.e. IP/UDP/RTP/PAYLOAD. Our proposed system requires the following new delay parameters: T_{arr} is the delay introduced by the waiting time to receive a second packet to perform the modulo two addition of the packets; T_{xor} is the time to compute the operation itself which is dependant on hardware processors and T_{ldata} is the time associated with larger packet data due to synchronisation that consists of T_{sync}, the delay introduced by additional header for synchronisation and T_{data}. Equation (5.4) defines the values for a single standard packet, T_p, and a single network coding packet, T_{ntw} [78].

$$\begin{aligned} T_p &= T_{oh} + T_{data} \\ T_{ntw} &= T_{arr} + T_{xor} + T_{oh} + T_{data} + T_{sync} \end{aligned} \tag{5.4}$$

Two types of delays can be seen from Figure 5.5: fixed and variable. Overhead and data delays are related to standard specification and do not vary within a VoIP session. Similarly T_{xor} depends upon the hardware chosen for the encoder system which without loss of generality can be considered a fixed delay. Conversely delay introduced by network coding varies upon T_{arr} and it is derived from the design of our proposed system which maximises the probability of performing network coding. The system is based on two independent queues which theoretically receive packets at average time equal to the packetization interval time. Consider that the arrival of a packet is modelled as two independent Poisson processes with a single service time. From queueing theory it is known that two joined Poisson processes is also a Poisson processes [80]. Therefore, since the minimum number of received packets has to be two, one packet per station, our queueing design can be modelled as follows [80]:

$$P_2(t_w) = \frac{(\lambda t_w)^2}{2} e^{-\lambda t} \tag{5.5}$$

where the maximun probability of having two packets is defined as $\frac{dP_2(t_w)}{dt} = 0$ and calculated as

$$\frac{dP_2(t_w)}{dt} = \frac{\lambda^2}{2}\left(2te^{-\lambda t_w} - \lambda t^2 e^{-\lambda t_w}\right) \tag{5.6}$$

the equation has a solution of

$$t_w = \frac{2}{\lambda} \tag{5.7}$$

In a generic Poisson process, λ is the average arrival time which is equivalent to the average of generating one packet per packetization time. Since the system is a joined Poisson process the average arrival time is now doubled, which by considering a commonly used 20 ms per packet packetization time gives $\lambda = 1/10$ ms. This concludes that the maximum probability of having two packets in the queue is every $t_w = 20$ ms. In other words, the minimum time that the server has to wait to maximise network coding performance is 20 ms which has been adopted in our implementation as seen in the next section. Although the waiting time for a network coding process in VoIP holds at most 20 ms, it does not mean that if two packets arrive one after each other with $T_{arr} \to 0$ the real delay introduced by network coding simply depends on T_{xor} and the extra overhead introduced by synchronisation. This is interesting because the capacity of a broadcast network changes if MD of a network coding system is lower than the MD of the conventional method. From equation (5.2) the capacity of the network can be derived as the number of sessions that can be placed in a packetization interval.

$$C_{calls} = \frac{T_g}{T_t} \tag{5.8}$$

Many papers including [81], [82], [83], [84] and [79] used equation (5.8) to measure the capacity a network utilising the IEEE 802.11 protocol. Although this formula can be used to any type of network, these papers always address the capacity of the wireless channel to host VoIP session based on a point-to-point topology from a wire to wireless network, similar to a peer-to-peer system. Conversely in this chapter, we focus on a star-based protocol which changes the values we introduce for T_t. If the network is half duplex there is a threshold to reach in order to be more efficient than the conventional method and this is shown in next theorem [78].

***Theorem* 1.** *A VoIP single packet's delay has to be greater than the delay introduced by network coding in order to ensure a feasible VoIP communication with network coding schemes.*

$$T_p > T_{arr} + T_{xor} + T_{sync} \tag{5.9}$$

In order to corroborate our method we solve the equations for IEEE 802.11 [85], a half duplex wireless broadcast network and the results are compared to a full duplex network such as IEEE 802.3 [86]. Since different type of codecs result in different T_{data} values, three audio codecs described in Section 2.2.3 on Page 36 are considered.

5.3.1 Minimum Delay for IEEE 802.11b

The link layer at IEEE 802.11 is based on a Carrier Sense Multiple Access protocol with Collision Avoidance (CSMA/CA). This protocol is designed for networks where sensing the network and sending information at the same time is not compatible which is the case of wireless networks. The standard defines a binary exponential backoff method named Distributed Coordination Function (DCF) which allows stations to transmit data units over a Contention Period (CP). Optionally a Point Coordination Function (PCF) is described as a guarantor of a minimum QoS per station. This method uses an AP to manage a Basic Service Set (BSS) in a round robin pooling system to synchronise with the stations and offers a Contention Free Period (CFP) to each and one of them. A VoIP Proxy server is connected via a wire to an AP that gives, in DCF mode, access to multiple stations where all nodes compete for the channel including the AP.

Priority access to the wireless medium is controlled by a time interval defined as Inter-Frame Space (IFS). Every sent frame has to wait at least a period of time before transmitting a message named Short Inter-frame Space (SIFS), T_{SIFS}. The next longest interval is the PCF Inter-Frame Space (PIFS), an interval designed to give priority to an access point over the next station at a BSS. At a lower priority, and therefore longer waiting period, it is defined as DCF Inter-Frame Space (DIFS),

T_{DIFS}. Next longer interval time is Arbitration Inter-Frame Space (AIFS) followed by the Extended Inter-Framed Space (EIFS).

One of the drawbacks of using wireless channels over centralised nodes is that exchanged packets between stations and AP might not be read by all stations arising the hidden station problem. IEEE 802.11 overcomes this issue by using Request To Sent (RTS) frame before any data is sent. RTS is confirmed by a Clear To Send (CTS) message followed by the data unit and the correspondent Acknowledgement (ACK) frame, T_{ACK}. Although this solution solves the problem it requires a large overhead. Instead, a basic frame exchange can be done in IEEE 802.11 so that no RTS or CTS packets are exchanged before each transmission cycle. This requires a timing interval protocol to maximise the usage of the channel diminishing the probabilities of a collision. An example of such transmission is illustrated in Figure 5.6. Any station willing to send a data frame has to sense the medium for DIFS period. If the channel is free the station will generate a random backoff period ($T_{backoff}$) additional to DIFS. In this case, $T_{backoff} = rand(0, CW_{min}) \cdot T_s$ where CW_{min} is the minimum contention window, $rand$ is a uniform random function between 0 and CW_{min} with an average of $CW_{min}/2$ and T_s refers to the slot time. If after the backoff time the channel is free data is transmitted. AP receives the frame and responds after a SIFS period with an ACK frame confirming the delivery of the frame. Meanwhile other stations such as *Bob*'s, generate a Network Allocation Vector (NAV) indicating the period the medium will be busy.

IEEE 802.11 physical overhead is defined as a computational of Physical Layer Convergence Protocol (PLCP) Data Unit (PPDU) as seen in the upper graph of Figure 5.6. The preamble is required at the receiver side for synchronization purposes and it comprises a set of synchronisation bits plus a Start of Frame Delimiter (SFD). There are two optional preamble sizes: either short with 56 bit or long with 128 but in both cases the preamble will be sent at 1 Mbps. PLCP header consists of 48 bits that allows the receiver to know the physical specification of the frame such as modulation type and it is vendor's choice to choose the data rate which in our case is set to 2 Mbps. The rest of the headers are sent at the rate agreed between AP and stations which we define as R_{AP} and is set to 11 Mbps. The frame control provides the specifications of sent frame, duration is a field of 2 bytes that includes Frame Check Sequence (FCS) and Address 1 is referred as the source address. These fields compose the minimal frame size. Address 2, 3 and 4 refer to the destination addresses,

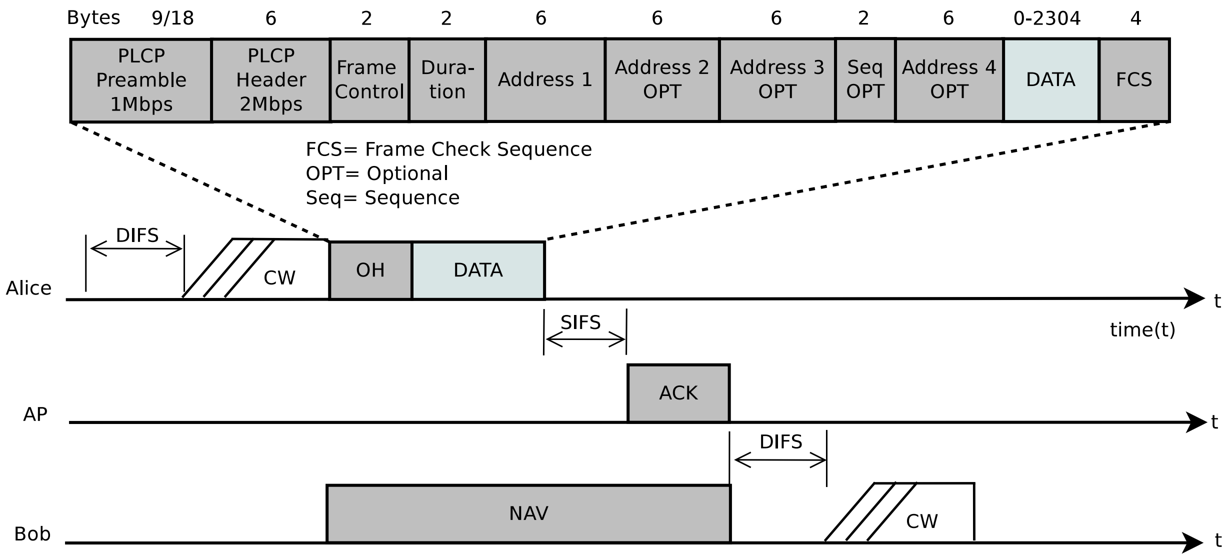


Figure 5.6: Transmission of a packet in IEEE 802.11 [85].

transmitting station and receiving station respectively but these are optional. Finally the sequence number is associated to the MAC Service Data Unit (MSDU).

To compute MD and considering the above description of IEEE 802.11, we define the delay introduced by this overhead as follows [78]

$$\overline{T_{oh}} = T_{DIFS} + T_{CW} + T_{header} + T_{SIFS} + T_{ACK} \tag{5.10}$$

where T_{header} is the time interval of PLCP preamble at 1 Mbps, PLCP header at 2 Mbps and the rest of the frame header at the data rate agreed between station and AP i.e R_{AP}. T_{ACK} is a 14 byte frame transmitted at 2 Mbps and T_{CW} is the average time before any packet is sent. In this case, the waiting time introduced by the contention window is compulsory for all the peers and therefore its average value is considered. Hence, definition of the time delay introduced by IEEE 802.11 can be read as follows considering $T_{arr} \rightarrow 0$, R_{AP} in *bits per second* and $B_{payload}$ and B_{sync} in bytes [78]:

$$\begin{aligned} T_t &= (2 \cdot T_p) + T_{ntw}, \text{ from (5.4)} \\ &= 2 \cdot (\overline{T_{oh}} + T_{data}) + \\ &\quad (T_{arr} + T_{xor} + \overline{T_{oh}} + T_{data} + T_{sync}) \\ &= 3 \cdot (\overline{T_{oh}} + T_{data}) + (T_{sync} + T_{xor}) \end{aligned}$$

Table 5.1: Parameter values for IEEE 802.11b

Parameters	Time (μs)	Size (bytes)
PLCP Preamble (short)	72.00	9
PLCP Header	24.00	6
MAC Header + FCS	20.36	28
ACK	56	14
SIFS	10.00	
DIFS	50.00	
Slot	20	
CW_{min}	31 slots	

$$= \quad 3 \cdot \overline{T_{oh}} + \frac{8}{R_{AP}} \left(3B_{payload} + B_{sync}\right) + T_{xor} \tag{5.11}$$

which means that the minimum delay is not only dependant on the data rate of the network but a fixed delay is introduced by the MAC layer, highlighting the limitations that IEEE 802.11 has when sending packets with an increasing throughput. A summary of utilised IEEE 802.11b parameters is presented in Table 5.1.

Table 5.2: Minimum Delay Bounds and Capacity for IEEE 802.11 [78].

-	G.711		GSM 06.10		Speex-mod4	
	CM	NC	CM	NC	CM	NC
T_t(ms)	2.751	2.067	2.382	1.790	2.367	1.779
C_{calls}	7.26	9.67	8.39	11.17	8.45	11.24

CM = Conventional Method, NC=Network Coding method

A comparison between conventional method and network coding is shown in Table 5.2 using values from Table 5.1. The assumption made in this method is that multicast networks provide an ACK frame after a packet is sent. According to [85] multicast networks do not conceive ACK frames due to obvious congestion if all receivers decide to acknowledge the delivery of the packet. Thus, the research presented in [87] proved that the use of a Leader Base Protocol (LBP) on the BSS outperforms conventional multicasting. The method works as follows; The AP sends a multicast RTS (m-RTS) that is only replied by the leader with an (m-CTS). The leader is selected according to first in first served method, in other words, the first station connected to the service set and willing to listen to multicast packets becomes the leader. Once the leader is chosen, packets are exchanged without RTS/CTS messages. Every packet received by the leader is acknowledged. In this chapter, the destination port of the multicast server is decided by the station that started the VoIP session. Equally this station is the only station sending acknowledgements back to the AP. This method ensures that the packet loss rate can be reduced to a minimum ensuring a good quality service. Alternatives to this multicasting method can be found in [88], however, most of the solutions are either for lower layers or upper layers of RTP.

Values at Table 5.2 present the theoretical boundaries for three audio codecs. As it can be seen, network coding in comparison to conventional method improves the number of VoIP calls a BSS can handle. G.711 can handle two more VoIP calls and the other two codecs, GSM and Speex, are improved by three VoIP calls. The main reason behind this improvement is the fact that the wireless channel acts as a half duplex channel, making the network coding method more efficient for real-time services.

5.3.2 Full Duplex Networks with Network Coding

The advantages of network coding along with our system have been explained in Section 2.4 on Page 50 , however it is important to put into contest real case scenarios where network coding is advantageous to network performance. IEEE 802.11b is a broadcast wireless network that acts as a half duplex channel where clearly network coding can overcome some of the maximum flow graph problems, however this is not applicable to all broadcast networks. Full duplex networks perform as shown in Figure 5.7. In this scenario, packets between the SIP proxy and *Alice* can be exchanged at the same time as network coding does. The example shows *Alice* sending a message to the SIP proxy at t_0. In the next time unit, *Bob* sends a packet to the SIP proxy but now it is also used by the SIP server to send a packet to *Bob*. This is only possible if *Bob* and *Alice* have a full duplex channel. Equally in the next time frame, t_2, *Alice* receives a packet at the same time it sends a new packet to the proxy. This conventional method exchanges more packets than network coding and therefore is less efficient as is the case of IEEE 802.3 Ethernet connections. In conclusion full duplex networks do not have better performance in terms of time units if network

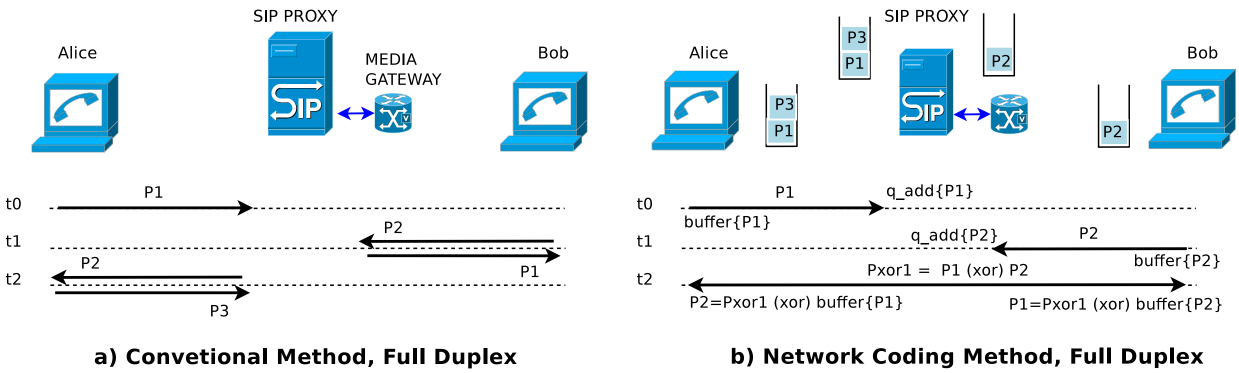


Figure 5.7: Design comparison between the conventional method and our proposed method with a full duplex channel [78].

coding is utilised.

5.4 Implementation of VoIP services with Network Coding

In this section, detailed specifications of the experiments are presented. Utilised hardware, queueing system parameters and delays related to the codecs are discussed in order to understand the performance of network coding over IEEE 802.11b.

Our proposed method has been carried out with a *D-link G604T* AP, a SIP proxy running a Fedora core 8 with Linux kernel 2.6.26 and a laptop with Ubuntu OS, Linux kernel 2.6.31 running multiple UA, i.e VoIP users. The AP includes 4 Ethernet ports and a IEEE 802.11b compatible wireless card that has been set to transmit with full power, sending beacon frames every 100 ms and RTS/CTS are enabled only if packet size exceeds 4096 bits. With this set-up, RTS/CTS are never sent and introduced overhead by beacon frames can be dismissed. The distance between stations has been kept in a range of 5 meters and therefore no major interference is considered. The SIP proxy station has two Ethernet cards whereas the laptop is provided with four cards; two Ethernet cards and two wireless cards that give us access to low level frame spoofing. SIP proxy and RTP clients are written in C code using *gcc* compiler.

Analysis of packets is carried out using Tcpdump [89] in monitor mode to capture packets as well as internal timers at each station. Capacity is measured capturing end-to-end delay of packets. This is possible due to the fact that each station has a dual interface and thus the same C programming process can capture sent and received packets from different network cards. Sequence numbers are used to confirm the packet loss as well as end-to-end delay. All phone calls have been performed over 20 times with a duration of 60 seconds and presented results are averaged values of the experimental results unless otherwise stated. Note that during CBR phone calls, the length of the conversation does not have any influence in throughput.

The system is based on a SIP Proxy connected to an AP through an IEEE 802.3 (Ethernet) 100 Mbps link and a wireless station connected to the BSS at a rate of 11 Mbps. Each VoIP call, referred as a *pair* of users since each VoIP call requires two UAs, generates one sender and two receivers *threads* at the SIP proxy and two senders with two receivers at the mobile station. Note that the UA's receiver is a multicast receiver joined to the corresponding IP and port number. The receivers are joined to the multicasting service on a single IP and an odd number of port destination starting at 7861 which is incremented by four each time a new VoIP call is generated. Other parameters in our experiments are shown in Table 5.1.

5.4.1 Implementation Setup

Queueing design has been based on the theoretical result obtained from equation (5.7) and its pseudo algorithm is shown in Algorithm 5.1. The waiting time that ensures the highest probability of having received one packet per station station is 20 ms, therefore the queue system has to satisfy $T_{limit} = q_{min} \cdot (20 \text{ ms})$ where q_{min} defines the minimum number of packets that has to be in the queue to perform the modulo two addition, $q_{min} \in \mathbb{N}$. This value ensures that most of the packets in the queues are *xor*-red. When a VoIP session is started the SIP proxy allocates two queues shown as *Alice*'s queue , *Q1*, and *Bob*'s queue, *Q2*. The queues are standard First In First Out queues where each time *qget* function is called the packet in the first position of the queue is retrieved. The algorithm sets a timer, T_{max}, and checks whether *Alice*'s queue has as many packets as q_{min}. If this is true, *Bob*'s queue is checked and if both queues contain more packets than q_{min}, they are sent by multicast *xor*-ing the payloads. If not enough packets have been received by *Bob*'s queue then the timer is checked. On the contrary, if *Alice*'s queue has not received enough packets *Bob*'s queue is checked and if the timer is over, $T_{max} > T_{min}$, it will empty the queue by multicasting the packets, this is the case where *Alice*'s sequence number is set to null.

It must be noted that in this algorithm q_{min} is not specified as different values achieve different performances.

Capacity measurements have been conducted for the wireless channel but a clarifying experiment for a full duplex channel is also carried out to point out that delay-wise the performance of network coding does not improved conventional methods, as seen in the next section. Capacity values have been obtained by generating consecutive threads in the wireless station. The fact that a single laptop is used reduces the chances of packet collision since only two stations at the BSS compete for the channel. Therefore the saturation limit of the network is a good reflection of networks maximum throughput.

5.5 Experimental Results for VoIP with Network Coding

Three main aspects are discussed in this section, throughput gain of network coding, efficiency of our proposed queueing system and delay limits calculated as described in the previous section. Firstly, a single VoIP session is captured and analysed to corroborate that network coding is feasible, i.e. equation (5.9) can be achieved. Secondly, the capacity of network coding is shown and discussed.

5.5.1 Feasibility of Network Coding

The network coding throughput gain is clearly shown in Figure 5.8 on a single VoIP phone call of 15 seconds where $q_{min} = 1$. This illustration corroborates equation (5.1) and confirms that in terms of throughput, the gain in network coding is of 50% if downlink interface is considered which is represented by the filtered results from the VoIP calls. The spikes shown every 5 seconds are the RTCP packet required by the standard. It is important to understand that the throughput performance is shown in Packets/second which is independent of the link layer and audio codec utilised.

Throughput performance also reveals the efficiency of our proposed method. As mentioned in the previous section q_{min} can be varied to achieve higher or lower efficiency in network coding, where efficiency is defined as the number of packets received with an *xor* applied to the total number of packets received. Table 5.3 shows the efficiency of a single VoIP session in percentages for both Ethernet IEEE 802.3 and

Algorithm 5.1 Network Coding Decision

```
T_limit = q_min · (20ms)
StartTime(T_max)
while 1 do
  ▷ Check Queue for Alice
  if Q1 ≥ q_min then
    ▷ Check Queue for Bob
    if Q2 ≥ q_min then
      while ((Q2 >= q_min)and(Q1 >= q_min)) do
        p1 = qget(Q1)
        p2 = qget(Q2)
        send(p1(xor)p2)
      end while
      StopTime(T_max)
      StartTime(T_max)
    else
      if T_max > T_limit then
        while Q1 ≥ q_min do
          p1 = qget(Q1)
          send(p1)
        end while
        StopTime(T_max)
        StartTime(T_max)
      end if
    end if
  else
    if ((Q2 >= q_min)and(T_max > T_limit)) then
      while Q2 ≥ q_min do
        p2 = qget(Q1)
        send(p2)
      end while
      StopTime(T_max)
      StartTime(T_max)
    end if
  end if
end while
```

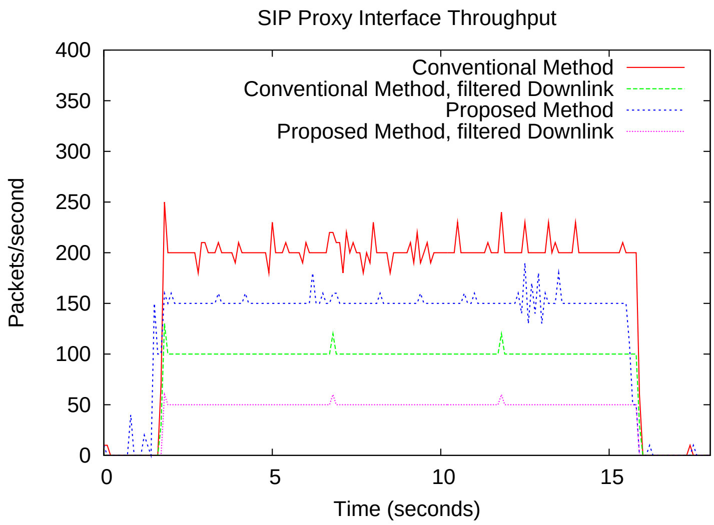


Figure 5.8: Throughput Performance Comparison [78].

Table 5.3: Efficiency of Network Coding Decision (NCD) % [78].

T_{limit}(ms)	IEEE 802.3 NCD (%)			IEEE 802.11b NCD (%)		
	G.711	GSM	Speex	G.711	GSM	Speex
20	99.86	98.81	99.80	99.10	98.48	98.63
40	99.10	99.27	99.77	99.04	98.03	98.87

Wireless LAN IEEE 802.11b. It can be observed that with $q_{min} = 1$, $T_{limit} = 20$ ms, efficiency is already very close to 100%. In fact, as it is monitored which packets were not *xor*-ed it can be seen that 2% of packets sent unchanged are at the beginning and end of the experiment which is due to both stations start sending packets at the same time but with a slight difference. Another important fact is that there is no major difference between codecs which fortifies the robustness of the design.

The critical parameter for a VoIP session is the delay. High delays induce packet losses since queues have finite size and the variation of delay causes jitter. In both cases the perception of voice quality is distorted if any of these occurs. In our experiments UA's queues are as large as the entire conversation which leaves the AP as the only bottle neck of the system rather than the end users or SIP Proxy. Jitter is compensated by an internal buffer set to ten payload packets, however this has not been taken into account for an end-to-end delay. Obtained results are presented in Figure 5.9.

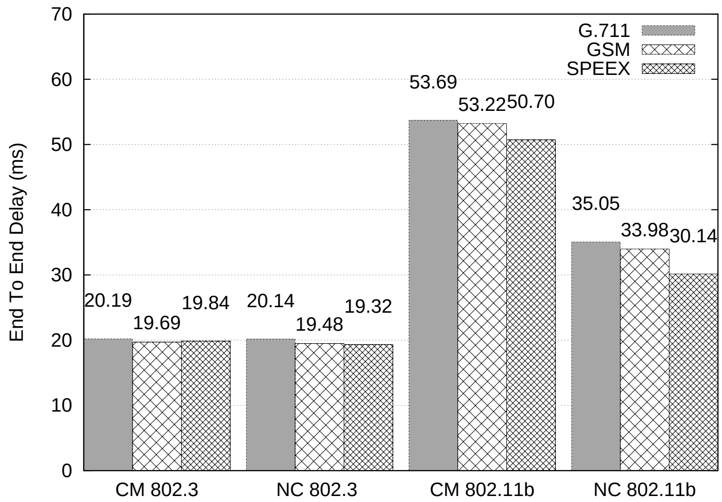


Figure 5.9: Delay comparison for a single VoIP system [78]. CM= Conventional Method, NC= Newtwork Coding method.

First two histograms refers to IEEE 802.3 Ethernet link layer results. Figure 5.7 forecasts that the delay comparison of network coding with conventional method is minimal in full duplex channels which is corroborated by first two histogram sets at Figure 5.9. In all cases, a single VoIP session's end-to-end delay is similar to the packetization delay T_g which is understandable since T_t values for the wire link are lower than a millisecond. Although theoretically G.711 should always have larger delay values, the slight variation within different codecs is considered insignificant since they all are very close to T_t. Conversely, last two histograms present the results for IEEE 802.11b link layer. These graphs prove one of the important achievements of this research where most importantly network coding does not only achieve better performance in terms of throughput but also in terms of delay. This difference between network coding and conventional method can be clearly seen by the delay difference, where the average delay of G.711 codec varies from 53.69 to 35.05. Equally the differences between the rest of the codecs change accordingly which corroborates the robustness of the method.

The average value of end-to-end delay is not meaningful if the inter-arrival time is not under certain levels. The inter-arrival time is defined as the time between the arrival of two consecutive packets. Conventional methods have an inter arrival of 20 ms since the packetization interval is 20 ms, however the performance of network coding has not been studied yet.

Figure 5.10 illustrates the inter arrival time of three randomly chosen phone calls, for both IEEE 802.3 and IEEE 802.11b channels. The full duplex channel results, left column, show a very steady delay values averaged to 20 ms. Each graph shows an inner graph with a higher resolution to give a full understanding of the performance. In reality, since the physical layer at IEEE 802.3 can sense the network while transmitting it does not have to wait until the network is idle. This explains the steady response in inter-arrival time and can be extended to any codec type.

The half duplex channel, right column, show a slight different panorama. From illustrations it can be seen that the average performance is 20 ms but maximum values are 112, 100 and 89 milliseconds for Speex, GSM and G.711 respectively. Since the channel acquisition is preceded by a randomly generated Contention Window (see equation (5.10)) the end-to-end packet delivery depends upon three times randomly generated value (one per station plus the response from the proxy). In addition the propagation delay introduced by the wireless channel has a negative effect on the performance in comparison to a wired channel. The inner pictures of each figure show a closer plot where the variation of such delay can be seen to fluctuate from almost 0 ms up to 40 ms. This is an expected variation since the SIP server has to wait for two packets before performing the joint packet. The high variation of the inter-arrival time compromises the receiver where jitter buffers have to be increased to achieve good quality of service. However, results clearly demonstrate that high peaks appear occasionally and do not dominate the general performance.

The proposed design confirms that Network Coding is a feasible method even for real-time services and unveils the need of a large jitter buffer to overcome the invariance introduced by the combination of wireless channel access and network coding itself.

5.5.2 Capacity of Network Coding on IEEE 802.11b

The comparison of capacity measurements for IEEE 802.11b wireless network with Conventional Method (CM) and Network Coding (NC) method are presented in this subsection. The theoretical results summarised in Table 5.2 asserts that conventional method can handle up to 7 VoIP users for G.711 and 8 for GSM and Speex codecs. Figure 5.11 (a) shows our experimental measurements obtained with regards to the CM and NC method. The x axis represent the number of VoIP pair of users (recall

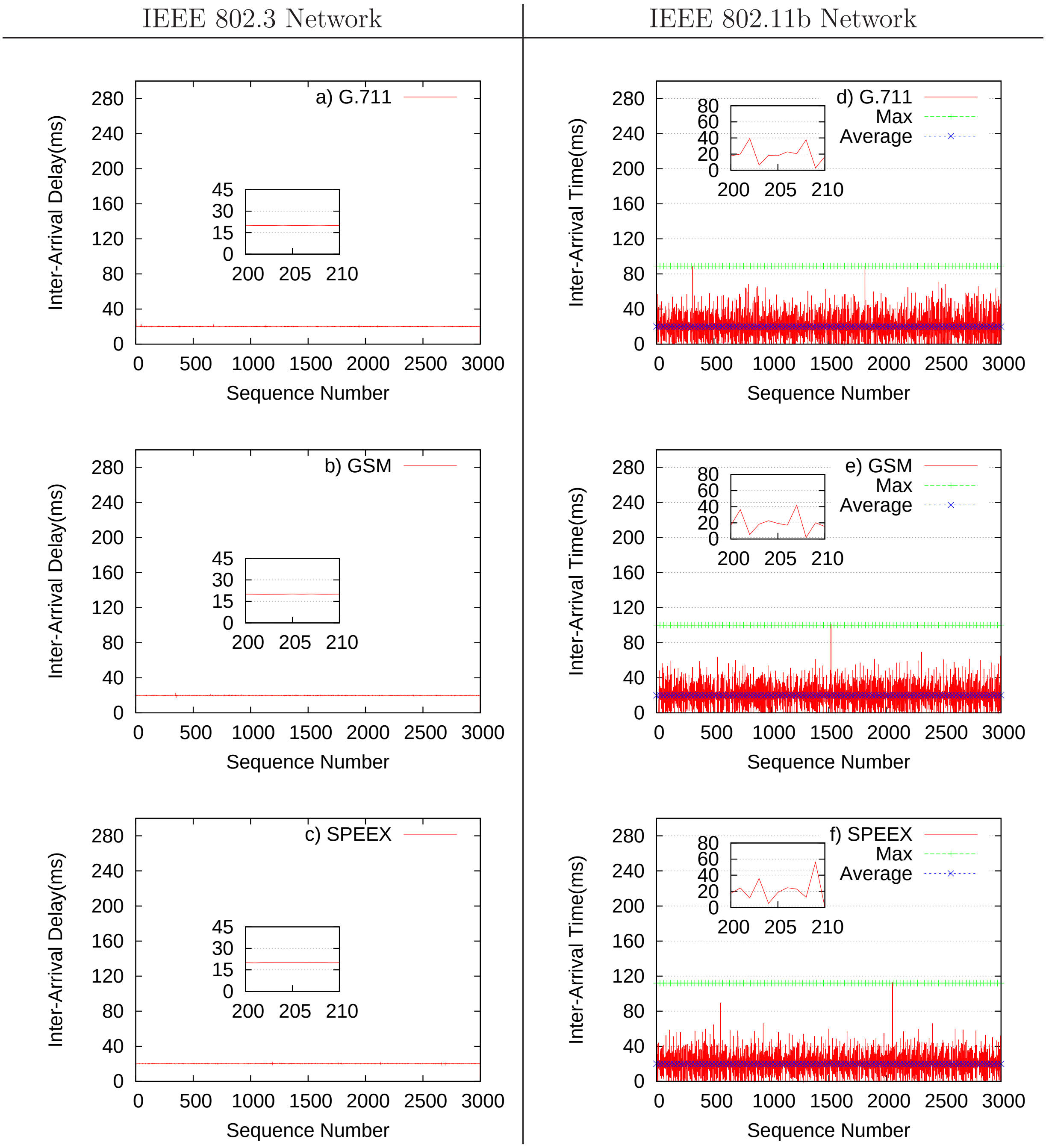


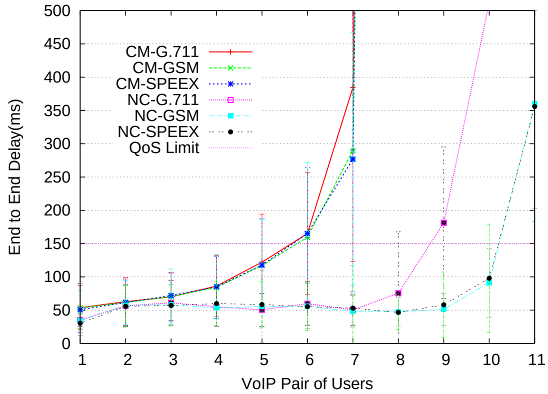
Figure 5.10: Inter-Arrival Delay for Network Coding over IEEE 802.3, full duplex channel (left column), and over IEEE 802.11b, half duplex channel, right column

that a session involves two VoIP users) and y axis shows the end-to-end delay values. Each value has its standard deviation plotted along giving a full understanding of the overall performance of the phone call. The illustration also shows an ideal maximum delay given by the quality of service requirements stated by the E-model [2] for VoIP services. If this delay limit of 150 ms is taken as a guide line to determine whether the number of VoIP sessions are feasible or not, then CM experimental results show that G.711 GSM and Speex can achieve 6 VoIP pair of users. Comparing these results with the theoretical bound in Table 5.2, it can be seen that G.711, GSM and Speex are expected to achieve 7,8 and 8 VoIP calls respectively. This is understandable because the theoretical bound assumes an ideal scenario where there is no collision at all, i.e. a perfect channel. Nevertheless, as the number of pair user increases G.711 shows a higher delay than GSM and Speex as verified by the theoretical analysis.

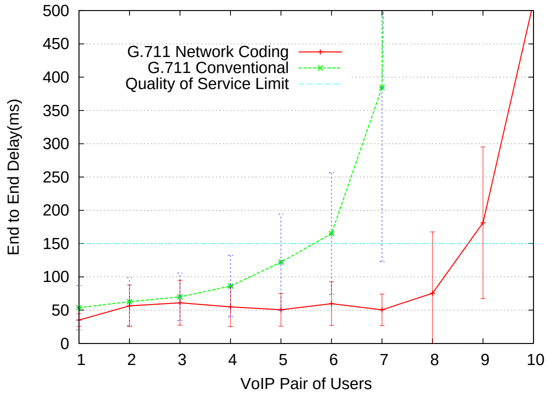
The NC performance depicts capacity values of 9 for G.711 codec and 10 for GSM and Speex. If these values are compared with the Table 5.2 where the theoretical boundaries are 9, 11 and 11 for G.711, GSM and Speex respectively, it can be seen that there is a close match between the theory and experimental measurements. In fact, network coding experiments are arguably a closer match to the theory than conventional method. We attribute this variation to the fact that network coding requires to access the channel 25 % less than the CM and therefore the performance of the model can achieve a better match with theoretical boundaries.

The flat characteristic of the overall performance of VoIP user pairs is a particular feature of the Internet Protocol and in general of broadcast networks. Recall that IP is designed for bursty load networks whereby users send a high load of information in a short period of time. This can also be understood if the broadcast network is compared to a water pipe, where the users traffic is the incoming water. The water flowing through the pipe is the same for any user as long as we do not exceed the maximum capacity of it, at which point the pipe is flooded. Figures in 5.11 are not any different. Delay performance is fairly stable until it reaches the boundaries of the link layer and delays increase significantly.

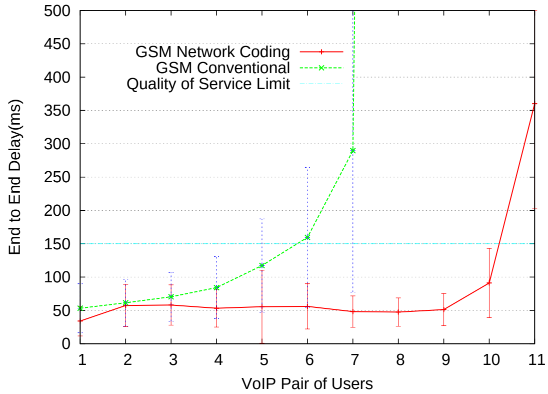
Illustrations from Figure 5.11 (b) to (d) are codec based comparisons, whereby G.711, GSM and Speex performance is contrasted individually for both conventional and network coding methods. Figure 5.11 (b) shows the comparison of the G.711 codec. In this graph, if strict delays below 150 ms are considered, it can be seen that 5 and 8 VoIP pair of users are the limits to CM and NC method respectively. This



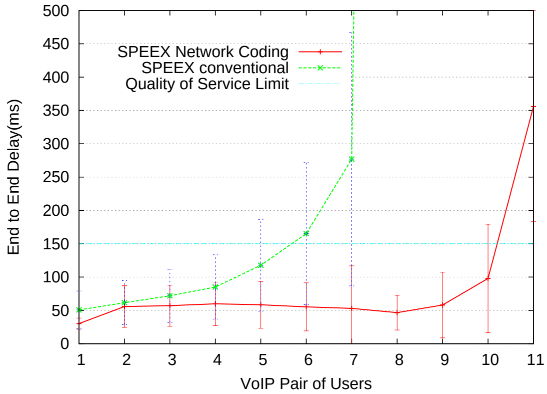
a) Full Results for CM and NC methods



b) Comparison of CM vs NC for G.711



c) Comparison of CM vs NC for GSM



d) Comparison of CM vs NC for Speex

Figure 5.11: Capacity results for IEEE 802.11b, where CM refers to Conventional Method and NC to Network Coding method [78].

proves that at least a 60 % of improvement over the conventional method is obtained which is considered a remarkable achievement. For the case of GSM, see Figure 5.11 (c), the strict difference between both methods proves to be 100 % of improved, a result that is also achieved for Speex codec as shown in Figure 5.11 (d).

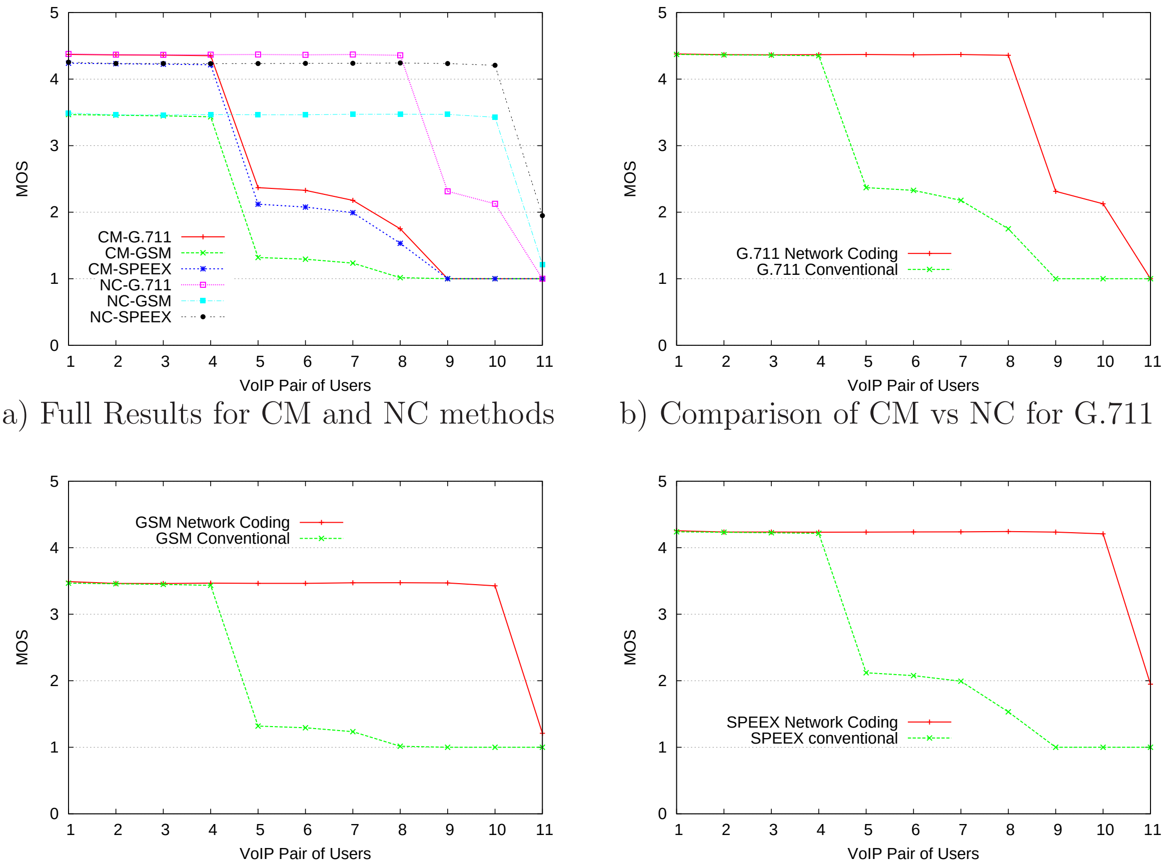
5.5.3 Mean Opinion Score for Network Coding

As mentioned in Section 2.3.2 on Page 45 VoIP call quality is often rated using the MOS method. Due to the limitations of this subjective method, the E-model is an elegant alternative to predict speech quality performance. In this section, the delay and packet loss results are utilised to calculate the *R-value* (see equation (2.4)) and transformed to the MOS value using equation (2.3). It is important to note that there are not packet losses in these experiments except last phone call of capacity where very large delays produce packet losses higher than one percent.

Figure 5.12 (a) illustrates the quality response for both conventional and network coding methods. In the conventional method the codecs in descending order of quality are: G.711, Speex and GSM. This is due to the fact that there are no lost packets in the experiments for G.711 which being a lossless audio codec, will always achieve the best voice quality under ideal circumstances. For lossy compression audio codecs with the CM, it is remarkable that the performance of Speex, a CELP based audio codec with a payload size five times smaller than G.711, still achieves a very good performance. In this figure, rather than having a steady loss of quality, as would be the case for an increasing loss of packets, quality is kept stable until a breaking point (5^{th} phone call) where the QoS has dropped to minimal values for a VoIP call and further values are considered unsatisfactory. Results for NC method clearly show the better performance in comparison to CM. The breaking point is at 8^{th} phone call for G.711 and 11^{th} for Speex and GSM, standing out as the better choice of compressing audio codecs for higher capacity, a feature not seen in the conventional method due to the nature of IEEE 802.11b medium access time delay.

Figure 5.12 (b) to (d) depict the comparison of each codec individually. From these illustrations, it can be seen that network coding is a better choice with any of the audio codecs chosen. In Figure 5.12 (b), G.711's difference between breaking points can be considered 9 to 5 for NC and the CM respectively, where VoIP QoS is improved 100%. The GSM performance, see Figure 5.12 (c), where the breaking points are 5 to 11 shows a 125 % improvement which is also the case for Speex as seen in Figure 5.12 (d). If experimental results are compared with the theoretical response it can be seen that overall performance is slightly below expected capacity measurements (see Table 5.2). This difference is believed to be the difference between real experiments and ideal theoretical response where collision free links are expected.

This study clearly clarifies that network coding for VoIP services not only is feasible but it improves the overall performance in throughput, delay and consequently QoS.



a) Full Results for CM and NC methods
b) Comparison of CM vs NC for G.711
c) Comparison of CM vs NC for GSM
d) Comparison of CM vs NC for Speex

Figure 5.12: Mean Opinion Score (MOS) results for IEEE 802.11b, where CM refers to Conventional Method and NC to Network Coding method [78].

5.6 Conclusion

Network coding introduces new opportunities to maximise the throughput performance adding an extra cost of computation. Combining VoIP services and network coding is an innovative idea to reduce the network load. In this chapter, an application layer coding method is presented to handle unicast VoIP calls where packet multicasting results in remarkable gains in throughput and delay over half duplex channels. The method involves a queuing system and synchronisation packet designed to be fully compliant with current VoIP services but satisfying QoS requirements at the same time. Experimental results are based on an extensively used wireless network standard, IEEE 802.11b, proving the trade-off between network coding and additional delays as seen in equation (5.9). Furthermore, a maximum 100% improvement of VoIP call throughput over conventional methods, as shown in Figure 5.11 (a), is evidence that network coding enhances network performance.

CHAPTER 6

Conclusion and Future Work

6.1 Conclusion

NEXT generation of VoIP communications will involve a mixture of current and future technologies where customer satisfaction will play a key role. As more technologies are developed the likelihood of using SIP proxy with a merged Media Gateway that interact within different networks increases substantially. The nature of this real-time service over heterogeneous networks depends on the performance that the Media Gateway can deliver. Current intermediate nodes simply *store and forward* the message to minimise end to end delay and packet loss probabilities. Although initially this method reduces the size of the queue at the node it is not always the best solution to guarantee specific QoS. Hence, in this thesis the SIP proxy with a merged Media Gateway is considered as an active entity aware of customer satisfaction where QoS performance is a priority of the network.

State of the art speech prediction algorithms do not consider heterogeneous networks and as a consequence of that each technology tackles QoS individually. In this thesis, VoIP performance is studied for a variety of network topologies and structures, based on real case scenarios implemented in hardware. The analysis of heterogeneous networks is divided in two parts: firstly, a SIP proxy with a merged Media Gateway with awareness of global speech quality is investigated to activate FEC systems that overcome with high packet losses; secondly, an new analytical model to predict speech quality over cascaded networks is researched and compared with customers quality perception. This model provides a fast assessment of speech quality over heterogeneous networks. In addition, a more efficient VoIP method based on network coding schemes is developed without dependency on the physical layer. This independence ensures that VoIP can take advantage of any broadcast network with minor modifications, maximising the throughput of current and future broadcast networks.

In Chapter 3, a SIP proxy with a merged Media Gateway is implemented with a new perspective of quality degradation in VoIP communications. In this chapter, based on a real case scenarios where VoIP is bridged to different technologies, end-to-end VoIP quality is monitored from the Media Gateway that connects both end users. As it is common for Media Gateways, each users sits in a different link layer with different audio codec characteristics which results in a performance that only the intermediate node can assess since it has access to both network features. Conventional methods have not considered a global VoIP performance but in this chapter the awareness of global QoS is considered to configure a FEC system avoiding single packet losses. The results shown in Figure 3.10 on page 78 not only asserts that FEC improves the performance but also proves that if the communication between end users is considered or modelled as a single network, the quality estimation of the session is clearly far from realistic values. The experiments from this chapter are investigated using the MOS test which demands a large number of people and time. Taking this into consideration a more efficient approach to predict VoIP performance under heterogeneous networks is required.

In Chapter 4, a new analytical model is presented to predict speech quality performance over n number of networks providing the network can be mathematically modelled by FSMC. The proposed model expands the limitations shown by the current E-model by two means: new audio codecs are measured and new formulae is introduced to cope with n networks. On one hand, GSM and SPEEX, in MOD 4, have been tested and analysed to obtained their I_e and Bpl parameters. On the other hand, the new formulae includes: a new impairment factor to equalise high packet losses and a definition of a new effective equipment impairment factor based on ρ factor that uses the I_e value of the channel. Experimental results based on two independent channels are carried out through the MOS test method and compared to the proposed analytical model where each network is modelled using FSMC channel models with $q = 0.7$ and $q = 0.1492$. The difference between the MOS and the analytical model showed low average error-margins between 1.68R and 8.35R.

If active GW's have a better quality performance at the cost of higher throughput then a more efficient approach with the same speech quality performance is desired. To achieve this, Chapter 5 proposes an active proxy that encodes packets. The new algorithm uses network coding schemes at the application layer to implement a new approach to VoIP communications over a single Basic Service Set (BSS), where unicast packets are encoded and multicasted to all users. The method requires a new payload header compliant with current protocols to achieve users synchronisation. The theoretical study developed for IEEE 802.11b predicates that there is a maximum waiting time for network coding to be performed or other wise queue sizes tend to increase infinitely. Considering this investigation, queue systems in the SIP proxy to encode packets are implemented where VoIP call throughput is increased by 100% for G.711 and 125% for GSM and Speex without undermining the quality of the phone call, a remarkable improvement from conventional methods.

In conclusion, this thesis shows evidences that active nodes in VoIP networks enhance both throughput efficiency and speech quality.

6.2 Future Work

The emergence of new technologies with different features is leading to a heterogeneous network where the end-to-end QoS has to be considered in order to satisfy the customer. Although the proposed algorithms in this thesis significantly facilitate the incorporation of two networks to a VoIP quality prediction there is still potential to further exploit these benefits. It should be noted however that the trade off between quality and throughput performance is an ongoing issue. Consequently the following points are viable future research ideas:

1. **E-model optimisation for further audio codecs and technologies:** The methods proposed in this thesis could be expanded further to a number of heterogeneous networks whereby cases such as wireless-wire-wireless communications are considered with wider audio codec range and more realistic wireless channel models.

2. **VoIP with network coding**. The potential of active SIP proxy GW has been demonstrated throughout this thesis to outperform conventional servers. A key feature of the proposed methods is to apply network coding schemes at the SIP proxy. This type of server can achieve up to 50% of throughput increase without loosing speech quality at all. This outstanding achievement has not been fully exploited because more complex coding algorithms and resource allocation techniques have not yet been applied.

Appendices

Appendix A

A Calculation of parameters for the E-model with default values

The Impairment factors R_0 and I_S are dependant on default values shown in Table 1, where abbreviations and units have been clarified. In this appendix, the formulae to calculate these values is described following the specification in [2].

Table 1: Default values and permitted ranges for parameters of the Emodel [2]

Parameter	**Abbr.**	**Unit**	**Default**	**Permitted range**
Send Loudness Rating	SLR	dB	+8	$0...+18$
Receive Loudness Rating	RLR	dB	+2	$-5...+14$
Sidetone Masking Rating	STMR	dB	15	10...20
Listener Sidetone Rating	LSTR	dB	18	13...23
D-Value of Telephone, Send Side	Ds	-	3	$-3...+3$
D-Value of Telephone, Receive Side	Dr	-	3	$-3...+3$
Talker Echo Loudness Rating	TELR	dB	65	5...65
Weighted Echo Path Loss	WEPL	dB	110	5...110
Mean One WayDelay of the Echo Path	T	ms	0	0...500
Round-Trip Delay in a 4-wire Loop	Tr	ms	0	0...1000
Absolute Delay in echo-free Connection	Ta	ms	0	0...500
Number of Quantisation Distortion Units	qdu	-	1	1...14
Equipment Impairment Factor	I_e	-	0	0...40
Packet Loss Robustness Factor	Bpl	-	1	1...40
Random Packet-Loss Probability	Ppl	%	0	0...20
Burst Radio	$BurstR$	-	1	1...2
Circuit Noise referred to 0 dBr-point	Nc	dBm0p	-70	$-80...-40$
Noise Floor at the Receive Side	$Nfor$	dBm0p	-64	-
Room Noise at the Send Side	Ps	dB(A)	35	35...85
Room Noise at the Receive Side	Pr	dB(A)	35	35...85
Advantage Factor	A	-	0	0...20

A.I Basic Signal To Noise Ration, R_0

R_0 is defined as [2]:

$$R_0 = 15 - 1.5(SLR + N_0) \tag{A.I}$$

where N_0 is the sum of noise sources N_c, N_{os}, N_{or} and N_{fo} , referred as all circuit noise powers, sender side's room noise, receiver side's room noise and noise floor at the receiver side, respectively [2]:

$$N_0 = 10\log_{10}\left(10^{\frac{N_c}{10}} + 10^{\frac{N_{os}}{10}} + 10^{\frac{N_{or}}{10}} + 10^{\frac{N_{fo}}{10}}\right) \tag{A.II}$$

$$N_{os} = Ps - SLR - Ds - 100 + 0.004(Ps - OLR - Ds - 14)^2 \tag{A.III}$$

$$N_{os} = RLR - 121 + Pre + 0.008(Pre - 35)^2 \tag{A.IV}$$

where Pre is the effective room noise defined as [2]

$$Pre = Pr + 10\log_{10}\left(1 + 10^{\frac{10-LSTR}{10}}\right)$$

$$N_{fo} = Nfor + RLR \tag{A.V}$$

A.II Simultaneous Impairment Factor, I_S

I_S is the sum of three impairment factors that happen simultaneously with the voice transmission [2].

$$I_S = Iolr + Ist + Iq \tag{A.VI}$$

where Iolr, Ist and Iq stand for the decrease in quality caused by too-low values of OLR, non-optimum sidetone, and quantising distortion, respectively. They are computed with the following equations [2]:

$$Iolr = 20\left[\left\{1 + \left(\frac{Xolr}{8}\right)^8\right\}^{\frac{1}{8}} - \frac{Xolr}{8}\right] \tag{A.VII}$$

where

$$Xolr = OLR + 0.2(64 + N_0 - RLR)N \tag{A.VIII}$$

$$Ist = 12\left[1+\left(\frac{STRM_0-13}{6}\right)^8\right]^{\frac{1}{8}} - 28\left[1+\left(\frac{STRM_0+1}{19.4}\right)^{35}\right]^{\frac{1}{35}}$$
$$-13\left[1+\left(\frac{STRM_0-3}{33}\right)^{13}\right]^{\frac{1}{13}} + 29 \tag{A.IX}$$

where

$$STMR_0 = -10\log_{10}\left(10^{\frac{-STMR}{10}} + e^{\frac{-T}{4}}10^{\frac{-TELR}{10}}\right) \tag{A.X}$$

$$Iq = 15\log_{10}\left(1 + 10^Y + 10^Z\right) \tag{A.XI}$$

where

$$Y = \frac{R0-100}{15} + \frac{46}{8.4} - \frac{G}{9} \tag{A.XII}$$

$$Z = \frac{46}{30} - \frac{G}{40} \tag{A.XIII}$$

$$G = 1.07 + 0.258Q + 0.0602Q^2 \tag{A.XIV}$$

$$Q = 37 - 15\log_{10}(qdu) \tag{A.XV}$$

A.III Default parameter values of the Emodel

Using formulae from previous subsection and the default values in Table 1, the R-factor can be reduced as follows where the calculation of equations are summarised in Table 2.

$$\begin{aligned} R &= R_0 - I_S - I_D - I_{e-eff} + A, \text{using values from Table 1} \\ R &= 94.74 - 1.41 - I_D - I_{e-eff} \\ R &= 93.33 - I_D - I_{e-eff} \end{aligned} \tag{A.XVI}$$

Table 2: Values of described equation in this appendix using default values from Table 1

Equation	Default Value	Unit	Reference
R_0	94.74	dBm0	A.I
N_0	-61.16	dBm0p	A.II
N_{os}	-75.74	dBm0p	A.III
N_{or}	-80.91	dBm0p	A.IV
Pre	38.01	dBm0p	A.V
Nfo	-62.00	dBm0p	A.V
I_S	1.41	dBm0	A.VI
$Iolr$	0.44	dBm0	A.VII
$Xolr$	10.17	dBm0	A.VIII
Ist	0.00	dBm0	A.IX
$STMR_0$	14.99	dBm0	A.X
Iq	0.97	dBm0	A.XI
Y	-5.21	-	A.XII
Z	-0.79	-	A.XIII
G	93.03	-	A.XIV
Q	37.00	-	A.XV

Appendix B

B Hardware Specification for the development of an embedded SIP proxy Gateway

The embedded SIP proxy Gateway (GW) is based on NGW100, a development board from AVR with an AT32AP7000 processor. The development platform has two Ethernet ports, a single serial port and several input/output pins. These pins are utilised in this development to connect the system to a GSM modem. The connection between the platform and the modem is carried out with a signalling system based on a RS-232 port and an audio codec with an integrated headphone power amplifier compliant with the Intel Audio Codec 97 specification, the CS4202. Figure 1 illustrates the hardware development for the SIP proxy GW connecting the processor board to self designed expansion board that manages the connection between the processor and the GSM modem. The figure is divided in three blocks specifying the processor board, the designed expansion board and the GSM modem. In each block, the main components described in this paragraph are highlighted. The expansion board is designed with Multisim software and the schematic of the entire development is divided in two figures. Figure 2 depicts the main schematic for the CS4202 whereas Figure 3 shows the RS232 connection for the modem control. The specifications of each component of the Printed Circuit Board (PCB) can be found in Table 3.

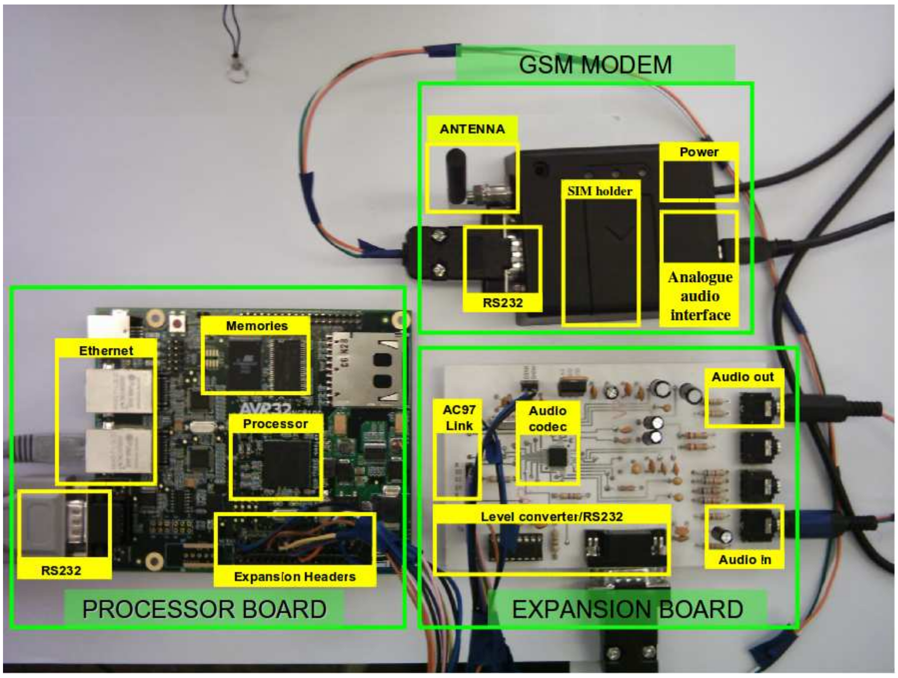


Figure 1: SIP Proxy GW hardware platform.

Table 3: Components of designed board

VALUE	SHAPE	N.	REFDES
1.5k	RES1300- 700X250	1	R16
1nF	KERKO5X4R5	6	C22,C23,C24,C25,C39,C40
1uF	ELKO5R5	2	C5,C26
1uF	KERKO5X4R5	1	C20
1uF	TANKO4˙5R5	1	C15
1uF	ELKO5R5	1	C27
1uF	TANKO4˙5R5	1	C14
2.2k	RES1300- 700X250	1	R17
2.2uF	TANKO4˙5R5	1	C19
6.8k	RES1300- 700X250	4	R10,R11,R12,R13
10k	RES1300- 700X250	2	R18,R19
10uF	ELKO5R5	3	C18,C37,C38
22pF	KERKO5X4R5	2	C1,C2
47	RES1300- 700X250	2	R1,R2
100	RES1300- 700X250	1	R25
100nF	KERKO5X4R5	13	C6,C7,C8,C17,C21,C30, C31,C32,C33,C34,C35,C36,C41
220k	RES1300- 700X250	2	R21,R22
220uF	ELKO5R5	2	C28,C29
470	RES1300- 700X250	2	R23,R24
AC97˙LINK	HDR1X5	1	U10
ANALOG Pwr.Sup.	HDR1X2	1	U8
CS4202b	TSQFP50-P900X	1	U2
DIGITAL PwrSup.	HDR1X2	1	U9
DSUB9F	DB9FL	1	J1
HC-49/US 25MHz	QUARZ˙HC49	1	X1
LM7805CT	TO220	1	U5
MAX3232	DIP16300	1	U7
Stereo Socket	3.5mm Stereo socket	4	U1,U3,U4,U6
USART	HDR1X5	1	U11

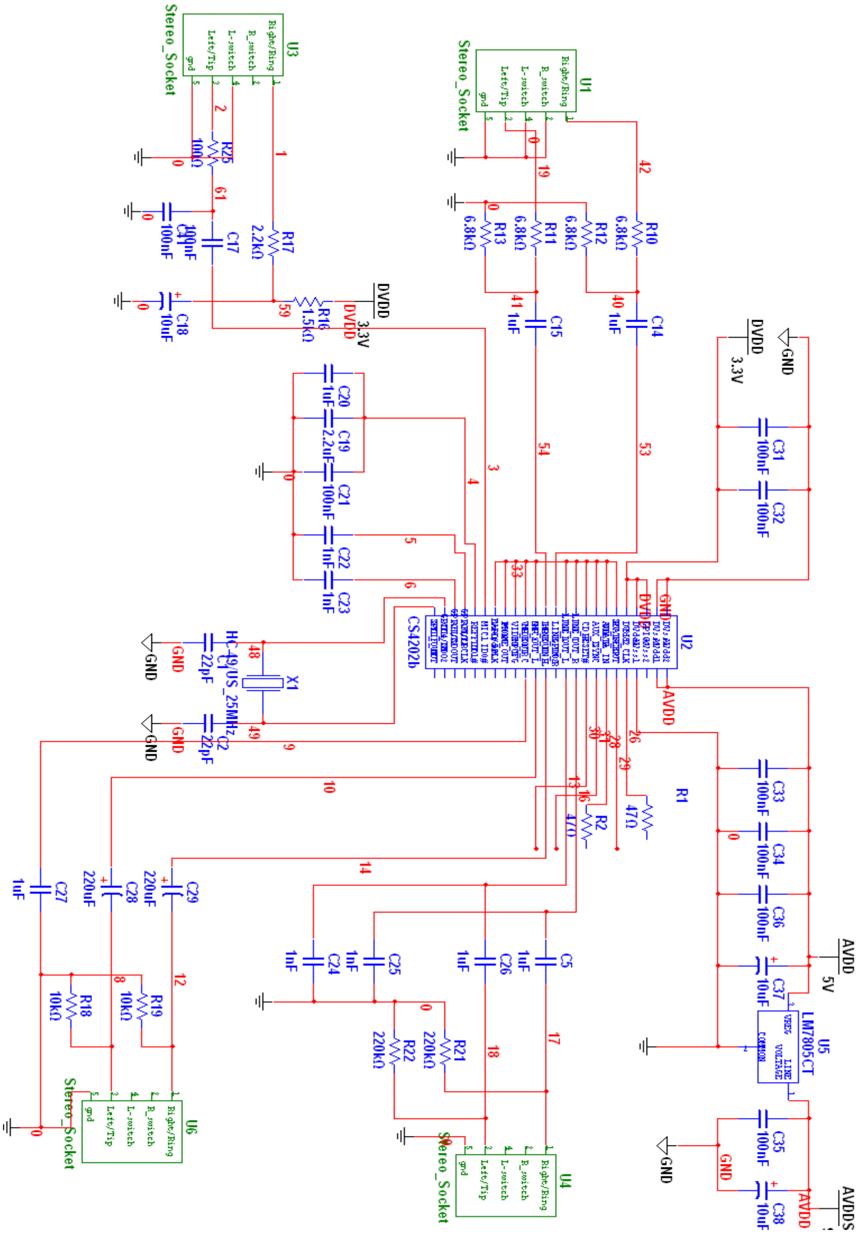


Figure 2: PCB Schematic Part 1, main board

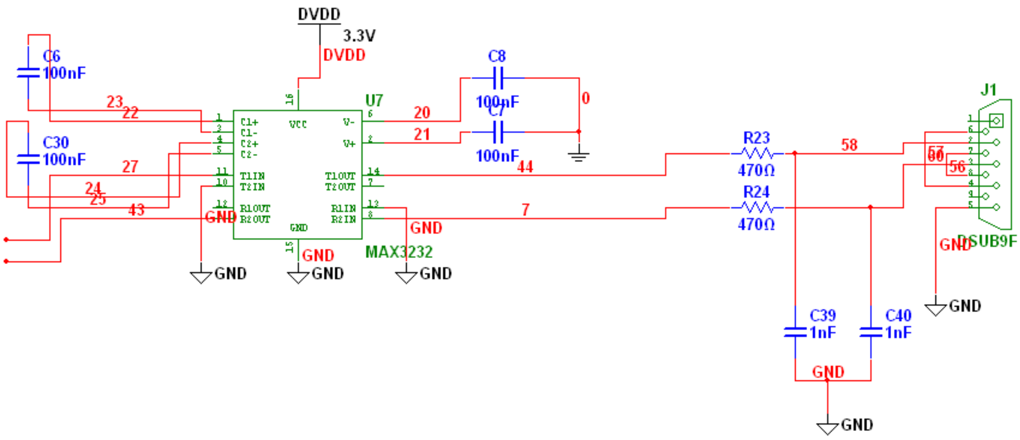


Figure 3: PCB Schematic Part 2, RS-232 serial Link

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