UNIVERSITI PUTRA MALAYSIA

EVALUATION OF VOICE TRUNKING PERFORMANCE USING AAL2 OVER ATM NETWORKS

SALIH MOHAMED S. KENSHIL

FK 2000 21

EVALUATION OF VOICE TRUNKING PERFORMANCE USING AAL2 OVER ATM NETWORKS

By

SALIH MOHAMED S. KENSHIL

Thesis Submitted in Fulfilment of the Requirements for the Degree of Master of Science in the Faculty of Engineering Universiti Putra Malaysia

June 2000

This humble thesis is dedicated to my ***late father***

for his endless urge for education

And

My family

For their encouragement and support throughout

The duration of my studies.

salmken@hotmail.com

Abstract of thesis presented to the Senate of Universiti Putra Malaysia in fulfilment of the requirements for the degree of Master of Science.

EVALUATION OF VOICE TRUNKING PERFORMANCE USING AAL2 OVER ATM NETWORKS

By

SALIH MOHAMED S. KENSHIL

June 2000

Chairman: **Associate Professor Borhanuddin Mohd. Ali, Ph.D.**

Faculty: **Engineering**

The urgent market needs for an efficient transport mechanism to carry voice traffic over Broadband Integrated Service Digital Networks (B-ISDN). Asynchronous Transfer Mode (ATM) network has been developed to integrate all types of traffic (e.g. voice, data and video) over B-ISDN networks. ATM Adaptation Layer 2 (AAL2) has been designed for efficient transport of voice, fax, and voice-band data traffic over an ATM virtual circuit. AAL2 allows the VBR sources to generate different packet sizes to be packed in an ATM cell, which increases the number of voice channels and decreases the packetization delay.

The growing trend of using ATM has also given great impetus to the use of AAL2 in wire-line trunking (e.g. PBX), which provides a good opportunity for multiplexing many voice channels over a single ATM virtual connection.

Voice trunking over ATM using AAL2 is the main subject of this thesis. The efficiency of AAL2 for voice transport in ATM networks is evaluated using a simulation approach. Additionally, the performance and capacity of an ATM multiplexer are analysed based on AAL2 technique. The power of AAL2 trunking and multiplexing high number of voice channels over a single ATM virtual connection makes it an efficient layer to support voice traffic with low delay and high efficiency using low bit rate as a reliable compression technique.

In this thesis, the comparison between AAL2 and AAL1 is made in terms of the number of users that can be carried over T1 ATM trunk. Then the focus is shifted to the use of AAL2 to support Voice and Telephony Over ATM (VTOA) networks. The effect of Timer-Composite Unit (Timer-CU) parameter, number of voice channels and voice coding rate on the packing density and link efficiency is further discussed. The results show that ATM cell packetization delay decreases when the number of users increases. Also, the Timer-CU affects the generation of partially filled cells when it is not set properly. Finally, the use of low coding bit rates to support high number of voice channels makes AAL2 a very efficient layer to carry VTOA networks.

Abstrak tesis yang dikemukakan kepada senat Universiti Putra Malaysia sebagai memenuhi keperluan untuk ijazah Master Sains.

PENILAIAN PRESTASI TRUNKING SUARA MENGGUNAKAN AAL2 MENERUSI RANGKAIAN ATM

Oleh

SALIH MOHAMED S. KENSHIL

Jun 2000

Pengerusi: **Profesor Madya Borhanuddin Mohd. Ali, Ph.D.**

Fakulti: **Kejuruteraan**

Ledakan pasaran masakini memerlukan mekanisma pengangkutan yang cekap untuk membawa suara menerusi Servis Rangkaian Digital Bersepadu Jalur Lebar (B-ISDN). Rangkaian Mod Penghantaran Tak Segerak (ATM) telah dibangunkan untuk menyepadukan kesemua bentuk trafik (contoh : suara, data dan video) menerusi rangkaian B-ISDN. Lapisan Penyesuaian ATM 2 (AAL2) telah dicipta untuk pengangkutan cekap bagi suara, fax, trafik data jalur-suara menerusi litar maya ATM. AAL2 membenarkan sumber Kadar Bit Boleh-ubah (VBR) untuk menjana pelbagai saiz paket untuk dibungkus dalam sel ATM, di mana ia dapat menambahkan jumlah saluran suara dan mengurangkan kelengahan pembungkusan.

Peningkatan trend untuk menggunakan ATM telah memberikan dorongan penggunaan AAL2 dalam jarak jauh talian-wayar (contoh : PBX), yang menyediakan peluang yang baik untuk memultipleks banyak saluran suara menerusi satu sambungan maya ATM.

Suara jarak jauh menerusi ATM menggunakan AAL2 adalah teras utama tesis ini. Kecekapan AAL2 untuk pengangkutan suara dalam rangkaian ATM telah dinilaikan menggunakan pendekatan simulasi. Tambahan pula, prestasi dan kapasiti pemultipleksan ATM telah dianalisis berdasarkan teknik AAL2. Kuasa Jarak jauh AAL2 dan pemultipleksan saluran suara yang tinggi menerusi satu sambungan maya menjadikannya lapisan cekap untuk menyokong trafik suara dengan lengah rendah dan kecekapan tinggi menggunakan kadar bit rendah sebagai teknik mampatan yang boleh dipercayai.

Dalam tesis ini, perbandingan antara AAL2 dan AAL1 adalah dibuat dari segi jumlah pengguna yang boleh dibawa menerusi penyaluran ATM T1. Kemudian tumpuan dialihkan ke AAL2 untuk menyokong suara dan rangkaian ATM menerusi telefon (VTOA). Kesan parameter Masa-Unit Rencaman (Timer-CU), jumlah saluran suara dan kadar pengkodan suara ke atas kepekaan bungkusan dan kecekapan sambungan akan dibincangkan. Hasil kajian menunjukkan lengah pembungkusan sel ATM berkurangan apabila jumlah pengguna bertambah. Juga Timer-CU memberi kesan ke atas penjanaan sel tak penuh apabila tidak dilaras dengan betul. Akhirnya, penggunaan pengekod kadar bit rendah untuk menyokong jumlah saluran suara yang tinggi menjadikan AAL2 lapisan yang cekap untuk membawa rangkaian VTOA.

ACKNOWLEDGEMENTS

First of all, I would like to express my greatest thanks and gratitude to **Allah** (S. W. T.) the most gracious and merciful for giving me the ability to carry out this work.

I would like to express my sincere appreciation and deep gratitude to Assoc. Prof. Dr. Borhanuddin Mohd. Ali (chairman of supervisory committee), who initially introduced me to the concept of ATM technology. Since then, he has provided me with many materials to search in and given me his wise counsel, guidance and valuable help. I am truly indebted to his unflagging support and encouragement.

I would also like to thank my supervisory committee, Dr. Sabira Khatun and Mr. Ashraf Gasim for their comments and suggestions were very helpful. I am very grateful for all of this assistance and untiring encouragement.

I am particularly grateful to who have helped me directly or indirectly in the preparation of this project. A special thanks also goes to the Libyan Education Ministry and Islamic Education, Scientific and Cultural Organization (ISESCO) for their unlimited support during my study that would never go unmentioned

Finally, I would deeply like to thank all of my family members for their moral, encouragement and support.

I certify that an Examination Committee met on 27 June, 2000 to conduct the final examination of Salih Mohamed S. Kenshil on his Master of Science thesis entitled "Evaluation of Voice Trunking Performance Using AAL2 over ATM Networks" in accordance with Universiti Pertanian Malaysia (Higher Degree) Act 1980 and Universiti Pertanian Malaysia (Higher Degree) Regulations 1981. The Committee recommends that the candidate be awarded the relevant degree. Members of the Examination Committee are as follows:

MD. MAHMUD HASAN, Ph.D.
Faculty of Engineering
Universiti Putra Malaysia
(Chairman)

BORHANUDDIN MOHD. ALI, Ph.D.
Associate Professor,
Faculty of Engineering
Universiti Putra Malaysia
(Member)

SABIRA KHATUN, Ph.D.
Faculty of Engineering
Universiti Putra Malaysia
(Member)

ASHRAF GASIM ELSID ABDALLAH, MSc.
Faculty of Engineering
Universiti Putra Malaysia
(Member)

MOHD. GHAZALI MOHAYIDIN, Ph.D.
Professor/Deputy Dean of Graduate School,
Universiti Putra Malaysia

Date: 30 JUN 2000

This thesis submitted to the Senate of Universiti Putra Malaysia and was accepted as fulfilment of the requirements for the degree of Master of Science.

KAMIS AWANG, Ph.D.
Associate Professor
Dean of Graduate School,
Universiti Putra Malaysia

Date: 13 JUL 2000

DECLARATION

I hereby declare that the thesis is based on my original work except for quotations and citations, which have been duly acknowledged. I also declare that it has not been previously or concurrently submitted for any other degree at UPM or other institutions.

Signed

Candidate.
Name: **Salih Mohamed S. Kenshil**

Date: 28 June 2000

TABLE OF CONTENTS

LIST OF TABLES

LIST OF FIGURES

LIST OF ABBREVIATINS

AAL1	ATM Adaptation Layer 1
AAL2	ATM Adaptation Layer 2
AAL3	ATM Adaptation Layer 3
AAL4	ATM Adaptation Layer 4
AAL5	ATM Adaptation Layer 5
AALs	ATM Adaptation Layers
ABR	Available Bit Rate
ADPCM	Adaptive Differential Pulse Code Modulation
AF-VTOA	ATM Forum-VTOA
ATM	Asynchronous Transfer Mode
B-CTI	Broadband-Computer Telephony Integration
B-ISDN	Broadband-Integrated Services Digital Network
BS	Base Station
BT	Burst Tolerance
B-TE	Broadband-Terminal Equipment
BTS	Base Transfer Station
CBEC	Cell Based Echo Canceller
CBR	Constant Bit Rate
CCITT	Consultative Committee on International Telecommunication and Telegraphy
CD	Cell Discarding
CDV	Cell Delay Variation
CDVAL	Cell Delay Variation Accumulation Length

CDVT	Cell Delay Variation Tolerance
CES	Circuit Emulation Service
CID	Channel Identifier
CLP	Cell Loss Priority
CLR	Cell Loss Ratio
CPE	Customer Premises Equipment
CPS	Common Part Sublayer
CS-ACELP	Conjugate Structure-Algebraic Code Exited Linear Prediction
CTD	Cell Transfer Delay
CTI	Computer Telephony Integration
DFQ	Delay Frame Queuing
DSI	Digital Speech Interpolation
EC	Echo Canceller
FCFO	First Come First Service
FEC	Forward Error Correction
GFC	General Flow Control
GSM	Global System Mobile
GSM-FR	GSM-Full Rate
GSM-HR	GSM-half Rate
HEC	Header Error Check
HEC	Head Error Control
IMT-2000	International Mobile Telecommunication-2000
IT	Internet Telephony
ITU-T	International Telecommunication Union-Telecommunication Standardization Sector

IWF	Inter-working Function
LAN	Local Area Network
LD-CELP	Low Delay-Code Excited Linear Prediction
LEO	Low Earth Orbit
LI	Length Indicator
LLC	Logical Link Connection
LSSGR	Local Switching System General Requirement
MC	Mobile Communication
MCR	Minimum Cell Rate
MEO	Mid Earth Orbit
MOS	Most Opinion Score
MSC	Mobile Switching Centre
N-ISDN	Narrowband-Integrated Service Digital Network
NNI	Network-to-Network Interface
nrt-VBR	non real time-VBR
OAM	Operation Administration and Management
PBX	Private Branch Exchange
PCM	Pulse Code Modulation
PCR	Peak Cell Ratio
PTI	Payload Type Identifier
QoS	Quality of Service
RM	Resource Management
rt-VBR	real time-VBR
SAD	Speech Activity Detection
SAF	Speech Activity Factor

SCR	Sustainable Cell Rate
SN	Sequence Number
SSCS	Service Specific Convergence Sublayer
STF	Start Field
STM	Synchronous Transfer Mode
SVC	Switching Virtual Circuit
TDM	Time Division Multiplexing
Timer-CU	Timer-Composite Unit
UBR	Unspecified Bit Rate
UNI	User-to-Network Interface
UUI	User-to-User Information
VBR	Variable Bit Rate
VCC	Virtual Channel Connection
VCI	Virtual Channel Identifier
VPI	Virtual Path Identifier
VPN	Virtual Path Network
VTOA	Voice and Telephony over ATM
WAN	Wide Area Network

CHAPTER 1

INTRODUCTION

1.1 Background

ATM stands for "Asynchronous Transfer Mode". It is primarily driven by telecommunication companies and is a proposed telecommunications standard for Broadband-Integrated Services Digital Network (B-ISDN). Development in current telecommunication industries indicates that ATM is the killing technology for the future B-ISDN. It provides bandwidth-on-demand at multimegabit-per-second peak rates, packet switched transport, traffic integration, cost effectiveness, and flexible data networking. ATM network can be viewed as integration of telecommunication networks (i.e. telephone network) and data communication networks (i.e. Internet). Hence, it satisfies all levels of media application requirements.

The telecommunications companies are investigating fibre optic cross the countries and cross the oceanic links with Gigabit/sec speeds. Also, they would like to carry in an integrated way, both real-time traffic voice and video, which have strict delay requirements, whereas in many data applications real-time delivery is not a primary concern. Even within delay-sensitive traffic (i.e., voice or video), different traffic streams may have different delay requirements, some data may contain more urgent information than others. Some traffic (i.e., data) is loss-sensitive and thus must be received without any errors.

ATM is capable to carry all traffic types over its adaptation layers AAL. In this project, we concentrate on voice telephony traffic than the others. In general, two kinds of methods can be used in transferring voice traffics in ATM networks, Constant Bit Rate (CBR) and Variable Bit Rate (VBR) to save the expensive transmission bandwidth. In second method, only the talkspurts are transmitted while the silences are removed. ATM Adaptation Layer type 1 (AAL1) is standardised to carry the coded voice information in ATM networks, voice is carried over AAL1 as constant bit rate traffic. AAL1 is suitable for transferring 64-kbps voice traffic, but it causes an unacceptable delay in cell assembly when it is applied to low-bit-rate voice traffic.

Since the trend now is more toward the use of low bit rates, the International Telecommunication Union (ITU-T) and the ATM Forum started discussing and eventually reached agreement on the use of ATM Adaptation Layer type 2 (AAL2) for low bit rate voice traffic.

In this thesis, AAL2 is used to reduce the cell assembly delay because it allows multiple voice channels to be carried in a single ATM cell. This scheme enables the payload of a cell to be filled quickly even when each channel has a low bit rate or to be sent even partially filled to avoid such delay for any packet. Under AAL2, the payload of a single cell consists of several short packets, called Common Part-Sublayer (CPS) packets, from different voice sources. These short packets decrease the packing delay, on the other hand, short packets decrease packing density and link efficiency, also increase the overheads on the system. The Timer-Composite Unit (CU) is used to control the cell-assembly delay and avoid any prolonged delay for any packet that causes degrading voice quality. Designs of AAL2 to be used for voice trunking are still under consideration and the final issues of specifications have not been standardised yet.

1.2 Voice over ATM

It is clear that telephone conversation is still and will remain the predominant form of human daily communications. Meanwhile, high-speed multimedia applications are growing rapidly in recent years, which push the existing networks to become B-ISDN. ATM considered as the core technology in building B-ISDN and ATM promises to be the key networking enables for multimedia applications, such as data, video, videoconferencing and voice.

Many solutions and technologies have been proposed for providing voice and telephony services over ATM. Examples are Circuit Emulation Service (CES), ATM telephone system, and signalling translation technique, etc. With recent technological advances and accompanying progress in standardisation of telecommunication interfaces, the promise of widespread availability of good quality packetized voice service should be fulfilled soon.

The ITU-T has already standardised the transport of CBR voice over an ATM packet network via an adaptation protocol AAL1. However, researchers are still looking for better ways in order to carry voice over ATM in good quality of services. Inexpensive, saving bandwidth and voice quality are the most important features to satisfy the critical need in the market for voice-over-ATM. The current standardisation efforts concentrates on the development of protocols, AAL2, that allow the multiplexing of packet streams from different VBR voice sources into a single ATM virtual circuit. The basic aspects of AAL2 are defined in ITU-T Recommendation I.363.2, and the voice aspects are defined in draft Recommendation I.366.2. AAL2 allows multiple channels to use a single cell; that is, a single cell consists of several short CPS-packets from different voice sources. These schemes are used to increase the link efficiency and the number of users in the system and to decrease the packetization delay of the ATM cells.

1.3 Problem Definition

The ATM protocol has become widely accepted for networks providing integrated multimedia services, including Voice and Telephony over ATM (VTOA) services. AAL1 is suitable for transferring 64-kbps voice traffic, but it causes unacceptable delay in cell assembly when it is applied to low bit rate voice traffic. For example, using AAL1, a 48-byte cell payload is filled in 6 ms at 64-kbps rate, but takes about 72 ms when a 5.3kbps voice codec is used. This 72 ms dose not include the propagation delay, queuing delay, etc, that the cell must undergo when it travels through the network.

When assessing the use of AAL1 for voice over ATM, it is important to note that the AAL1 protocol has the following limitations.

- Bandwidth is used even when there is no traffic.
- Multiple individual users are not supported on one ATM connection.

- Basic bit rate for voice transport is always 64kbps or bounds of 64kbps (Nx64).
- No standard mechanisms for compression, and silence detection and removal are supported.

However, although some ATM equipment manufacturers offered AAL1 as a standard based solution, it is not considered an optimum solution. In order to satisfy the critical need in the market for voice over ATM, AAL2 is defined in the ITU-T Recommendation I.363.2, designed with the capabilities to support multiple voice users on a single ATM connection and transport compressed/silence suppressed VBR voice traffic. Thus, in this project, AAL2 is used to support this kind of traffic with different source-coding rates and multiplexing high number of VBR voice sources traffic to be carried over a single ATM connection. Cell packing is controlled by Timer-CU parameter to avoid any prolonged delay that may happen to any packet received in an ATM cell.

1.4 Research Objectives

Since the unfinished standardisation process for voice over ATM, a lot of work has been done to carry VTOA. In this thesis, we use AAL2 to carry this type of traffic, and our objectives are as follows:

- To provide a useful and realistic evaluation of the efficiency of AAL2 for voice transport compared to AAL1
- To study how voice telephony can be carried in short packets using ATM Adaptation Layer type 2 (AAL2).