

## REAL-TIME AUDIO STREAMING SOFTWARE FOR LAN

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This report is submitted in partial fulfillment of the requirements for the  
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2005

## **DEDICATION**

To my beloved mother and family, friend, my supervisor Mr.Muhamad Syahrul  
Azhar b.Sani and people who support me directly or indirectly.....

## BORANG PENGESAHAN STATUS TESIS

JUDUL: REAL-TIME AUDIO STREAMING SOFTWARE FOR LAN

SESI PENGAJIAN: 2005

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I hereby declare that this project report entitled

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is written by me and is my own effort and that no part has been plagiarized without citations.

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(MR. MUHAMAD SYARUL AZHAR B.SANI)

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## ABSTRACT

Real-Time Audio Streaming software for LAN environment which is also known as RTAS is a audio streaming application which applied multicasting concept in distribution the streaming .The main problems that been identified when develop this project is that the current audio streaming applications such as ShoutCast, IceCast application using internet connection and cannot be applied in the LAN(Local Area Network) for the streaming. Shoutcast and IceCast also a WebBase application. To overcome these problems, this project is being carried to be applied in the LAN Environment only and do not required internet connection. Besides that, java programming language is applied in this application compare to visual basic because java programming can support both operating systems that are Windows platform and Linux platform. Many researches have been done regarding develop this application such as the multicasting concept, suitable codec, protocol such as RTP , compression and decompress technique and other which are related to this project. This project contains four modules such as Real-Time Audio Streaming Server/Client software, Real-Time Audio Streaming Channel and Request Streaming channel. Real-Time Audio Streaming Server is to display the requested client and server will send a welcome message to the client that they had join the server .Real-Time Audio Streaming Client to Client determine weather the server is alive or not and client will receive message from the server when they had join the server. Real-Time Audio Streaming channel is open the server RTP session for the streaming. Request Streaming channel for client to join the streaming session with click join.

## ABSTRAK

Project yang bakal dibangunkan adalah *Real-Time Audio Streaming software* juga dikenali sebagai RTAS suatu aplikasi yang menggunakan konsep *multicasting* dalam penghantaran *voice stream*. Masalah aplikasi yang sekarang yang kenal pasti dalam membangunkan aplikasi ini adalah, contoh aplikasi seperti Shoutcast, IceCast tidak boleh di aplikasi tanpa perhubungan internet. Aplikasi ini juga menggunakan *webbase* untuk *streaming*. Bagi mengatasi masalah berikut project ini telah dibangunkan untuk dia gunakan dalam LAN tanpa perhubungan internet. Dalam pada masa yang sama project ini juga mengaplikasikan *programming java* supaya project dapat diaplikasikan dalam platform *Windows* and *Linux*. Banyak *research* telah dijalankan bagi mencapai objective project in seperti konsep *multicasting*, *Codec* yang sesuai untuk *streaming voice*, penggunaan *Protocol* seperti RTP dan konsep *compression and decompress* teknik. Project in mengandungi 4 module iaitu *Real-Time Audio Streaming Server/Client software*, *Real-Time Streaming Channel* and *Request Streaming Channel*. *Real-Time Audio Streaming Server* akan memaparkan *Client* yang request stream dari server dan server akan menghantar welcome message kepada Client. *Real-Time Audio Streaming Client* akan menentukan sama ada server aktif atau tidak, jika ya client akan menerima message dari server. *Real-Time Streaming Channel* adalah server aktifkan *RTP session* untuk *streaming*. *Request streaming Channel* untuk client aktifkan *RTP session* untuk *join server*.

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## CHAPTER I

### INTRODUCTION

#### 1.1 Project Background

Behrouz A.Forouzan(2003) in the Local Area Network book said that streaming is sound and picture data flow in a digital stream form a server computer to your computer and read to hear in real-time, without having to download all of the content. Streaming media technology allows you to see or hear the content in just a few seconds, instead of having to wait for it. In the majority of cases streaming media is used to maintain control over the distribution of the media files because of it difficult to capture streaming video and audio.

The proposed projects that want to be carried out is about Audio Streaming concept where this project can solve many problem such as communication with two parties. To improve the quality Audio Streaming among community members by applying the (RTP) Real-Time Transport Protocol because this protocol for Audio Streaming because it's designed to address the needs for efficient delivery of streaming multimedia over IP. RTP protocol is the default protocol for streaming Windows Media. RTP protocol is also used for streaming ShoutCast/Real Video/Real Audio, streaming QuickTime video (.mov, .mp4, .sdp streams).



## **1.2 Problem statement**

There are many type of communication exist in this modern world such as sms,e-mail, message .Streaming is a new kind of communication that occur between two end users . Streaming is sound is a data flow in a digital stream form a server computer to your computer and read to hear in real-time, without having to download all of the content. Streaming media technology allows you to see or hear the content in just a few seconds, instead of having to wait for it. Below are factors that initiate the project:

### **1.2.1 Download Problem**

User have many problem when they tried to download the audio files or speech that already given .They have to spend many minute or hour waiting to downloading those audio files because those files is large size and haven been streamed .This can courses Internet traffic to the network and loss of data in media file transmission.

### **1.2.2 No being used proper Protocol**

There application uses UDP (User Data Protocol) and TCP (Transport Control Protocol) to transferring the media files but this is not a proper protocol to transfer media files because it can reduce the quality of media files. The proper protocol that suitable for transmitting the media files is RTSP(Real-Time Streaming Protocol) and Real-Time Transport Protocol.

#### **1.2.4 Poor Communication**

In some organization they have call every staff to attend a meeting or send e-mail to every staff if some announce, transfer data or important document because there no audio streaming. This kind of communication only wasting time, energy and money.

#### **1.2.5 Requires Internet connection and cannot use in LAN**

Most of the free current Streaming application cannot be used in LAN (Local Area Network) environment. So this application cannot be run without internet connection such as RealPlayer7, yahoo messenger, MSN messenger and so on. Before using this application they need to register and logon to the server and because of this matter, Streaming application need internet connection.

### **1.3 Objective**

- i. The main purposed of this project being carried out is to develop Real-Time Audio Streaming Software for the used in LAN (Local Area Network) environment. Organization such as KUTKM
- ii. This project can save time and money to the user because user can communicate through audio streaming and don't need to message,sms or e-mail. This project is an alternate way to deliver a message or a speech through audio stream
- iii. Used a suitable protocol for streaming Audio that is RTP (Real-Time Transport Protocol).
- iv. Only can be applied in the windows platform only.

## 1.4 Scopes

This Real-Time Audio Streaming is suitable for those whom in LAN (Local Area Network) environment and can be applied without internet connection. This developed application can support Windows only. So the modules for the project are stated below:

- i. The main purpose of developing the Real-Time Audio Streaming software to Stream audio speeches only.
- ii. This Audio Streaming can be used in the one-to-one or one-to-many communication.
- iii. Real-Time Audio Streaming Server –The requested client will be display and server will send a authentication message to the client that they had join the server.
- iv. Real-Time Audio Streaming Client– Client can determine weather the server is alive or not. Client will receive message from the server when they had join the server.
- v. Real-Time Audio Streaming channel– Server will open the RTP session for the streaming.
- vi. Request Streaming channel – Client will join the streaming session with click join.

## 1.5 Project significance

The target users for this project will be office staff in LAN environment. The user will communicate with each other with using the Real-Time Audio Streaming Software. This project can be used in the small area network such as LAN for communication in that LAN only. This software can be applied in Office side where the director can announce a issues on few seconds to his staff. The director doesn't need to send message, sms or e-mail to all his staff. Staff does not have to download. Announcement or a speech to hear the content .With this Real-Time Audio Streaming Software all the users can listen to the announce or a speech in real-time and it save Time and money. This made the organization save money and time for the information delivery that suit in the modern world. Thus, it will facilitate the user with provided optimal communication with end-user.

## 1.6 Expected output

The expected output of the project will have a front end system where the software can be install into the client computer. The front end application will install at the client computer and client can operate the application with mike, headphone and so on for communicate with other client .This application can communicated with the entire client in the same LAN.

With applied the RTP (Real-Time Transport Protocol) in this software to transmitting the audio speech without loses. RTP is most suitable protocol to transfer media data compares with other protocol such as UDP (User Data Protocol) and TCP(Transport Control Protocol). RTP takes advantage of streaming which breaks data into many packets sized according to the bandwidth available between client and server. When enough packets have been received by the client, the user's software can be playing one packet, decompressing another and downloading the

third. The user is able to start listening almost immediately without having to get the entire media file. Both live data feeds and stored clips can be the sources of data

## **1.7 Conclusion**

In the current market, there have many video conferencing applications which can use by users for many purposes depend to their needs. Before produce an application, some observations and survey on users about the application must be identified. All users' requirements regarding video conferencing application can be collected and can produce an application that can give the users more satisfied using this application. Peer to peer concept will play main role in this project and many advantages have been identified through doing research. So for the next activities, literature review and project methodology will take place. In literature section, search, collect, analyze and draw a conclusion from all debates and issues in relevant body of literature are made. The sources can get from books, journals, technical reports, web pages and others which related to this project. Project Methodology is a way to use all available technique, tools and approaches used to achieve predetermined objectives.

## **CHAPTER II**

### **LITERATURE REVIEW AND PROJECT METHODOLOGY**

#### **2.1 Introduction**

In this chapter 2, literature review presents the method of finding, collecting, analyzing and conclusion from all issues that raised in relevant field of literature. The research mostly done based upon the facts and findings on the internet sources and other research. The Literature review study is done from the books, technical reports, journals, and so on. The methodology being used is the System Development Life Cycle (SDLC) methodology approach where this SDLC have 5 main components that is System Investigation, System Analysis, System Design, System Implementation and System Maintenance & review.

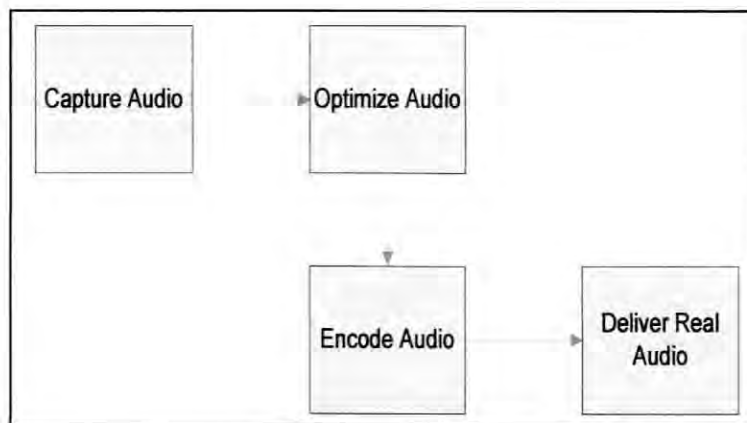
#### **2.2 Fact and finding**

There many researches has been done in order to develop audio streaming software such as RTP (Real-Time Transport Protocol), Audio Streaming delay, IceCast tools, Audio Codec and so no.

### 2.2.1 Real-Time Audio Streaming Protocol (RTSP)

Real-Time Audio Streaming Protocol by Juha HUUHTANEN Thompson (2003)[1] where multimedia application usually do not need the complexity of TCP and ,instead ,use a simpler transport framework. Most playback algorithms can tolerate missing data way better than long delays caused by retransmission. They don't require guaranteed in-sequence delivery either. Certain protocol has been developed to enhance the Internet and improved support of application like audio, video and interactive multimedia conferencing. The main protocols are the Real-Time Transport Protocol (RTP), Real-Time Control Protocol (RTCP), Resource Reservation Protocol (RSVP) and Real-Time Streaming Protocol (RTSP) Protocol.

RTSP takes advantage of streaming which breaks data into many packets sized according to the bandwidth available between client and server. When enough packets have been received by the client, the user's software can be playing one packet, decompressing another and downloading the third. The user is able to start listening almost immediately without having to get the entire media file. Both *live* data feeds and stored clips can be the sources of data. The Real Time Streaming Protocol is more of a framework than a protocol. It's meant to control multiple data delivery sessions, provide a way to choose delivery channels such as UDP, TCP and IP-multicast. The delivery mechanisms are based solely on RTP. RTSP has been designed to be on top of RTP to both control and deliver real-time content. Thus RTSP implementations will be able to take advantage of RTP improvements, such as RTP header compression. Although RTSP can be used with unicast, its use might help to smoothen the change from unicast to IP multicasting with RTP.



**Figure 2.1 Audio Streaming**

### 2.2.2 Real-Time Transport Protocol (RTP)

H. Schulzrinne(1996) Real-Time transport protocol (RTP), which provides end-to-end delivery services for data with real-time characteristics, such as interactive audio and video. Those services include payload type identification, sequence numbering, time stamping and delivery monitoring. Applications typically run RTP on top of UDP to make use of its multiplexing and checksum services; both protocols contribute parts of the transport protocol functionality. However, RTP may be used with other suitable underlying network or transport protocols .RTP supports data transfer to multiple destinations using multicast distribution if provided by the underlying network. Note that RTP itself does not provide any mechanism to ensure timely delivery or provide other quality-of-service guarantees, but relies on lower-layer services to do so. It does not guarantee delivery or prevent out-of-order delivery, nor does it assume that the underlying network is reliable and delivers packets in sequence. The sequence numbers included in RTP allow the receiver to reconstruct the sender's packet sequence, but sequence numbers might also be used to determine the proper location of a packet, for example in video decoding, without necessarily decoding packets in sequence.