

**AUDIO STREAMING SYSTEM USING REAL-TIME TRANSPORT PROTOCOL  
BASED ON JAVA MEDIA FRAMEWORK**

**By**

**IBRAHIM ASAAD AREF**

**Thesis Submitted to the School of Graduate Studies, Universiti Putra Malaysia in  
Partial Fulfilment of the Requirements for the Degree of Master of Science**

**March 2004**

To my Parents, Wife, Daughter,  
Brothers and Sister

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

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**March 2004**

**Chairman : Associate Professor Abdul Rahman Ramli, Ph.D.**

**Faculty : Engineering**

Audio streaming is an important component of multimedia networking applications. Today's Internet, however, offers only poor support for such streams due to the lack of the bandwidth and network traffic problems. The work presented in this thesis discusses the problems of real-time audio streaming and investigates solutions for improving the audio data transmitting over the network.

To achieve audio media data transmitting over the network in an efficient manner (real-time), the following issues: Initial delay of playing time (downloading time); current streaming protocols which can not cope well with network congestion; compression algorithms efficiency; network bandwidth utilization (network infrastructure); and security concerns of content owners, need to be considered.

In this thesis, the implementation method of a real-time audio streaming service system is discussed. The performance of the system implementation both in terms of resulting packet loss, initial delay and delay jitter is presented. This thesis describes audio streaming transmission protocols that are used to implement the system, the system architecture and how the system investigates and addresses the previous issues. A design proposal was outlined to provide an adaptive client/server approach to stream audio contents using Real-Time Transport Protocol (RTP) involving architecture based on the Java Media Framework (JMF) Application Programmable Interfaces (API). RTP protocol is the Internet-standard protocol for the transport of real-time data, including audio and video and can be implemented by using Java Media Framework (JMF). Java Media Framework library and the RTP protocol for audio transmission were used as development tools.

The developed system designed in this thesis together with experimental results proved that the system could be implemented successfully. A prototype of the developed system has been implemented and experiments over the Laboratory Local Area Network (LAN) and UPM campus LAN to investigate the issues mentioned before.

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

**SISTEM AUDIO STREAMING DENGAN MENGGUNAKAN REAL-TIME  
TRANSPORT PROTOCOL BERDASARKAN JAVA MEDIA FRAMEWORK**

Oleh

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Aliran audio adalah komponen penting dalam aplikasi jaringan multimedia. Internet hari ini tidak memberikan kemudahan yang terbaik utk tujuan aliran berkenaan disebabkan oleh kekurangan di segi lebarjalur dan masalah laluan jaringan. Berkaitan dengan itu, tesis ini cuba membincangkan masalah-masalah berkenaan masa-nyata aliran audio, dan mengenalpasti penyelesaian untuk memperbaiki kaedah penghantaran data melalui jaringan berkenaan.

Untuk penghantaran masa-nyata data secara berkesan melalui jaringan, pertimbangan terhadap beberapa fakto berkaitan adalah perlu, seperti kelewatan masa memainkan (masa muatturun), protocol yang sedia ada tidak mengambil kira kesesakan laluan jaringan, keberkesanan algoritma-algoritma pemadatan, penggunaan lebarjalur jaringan (infrastruktur jaringan), dan keselamatan maklumat yang dihantar.

Di dalam tesis ini, pelaksanaan kaedah untuk perkhidmatan masa-nyata aliran audio dibincangkan. Kemajuan pelaksanaan system ini dar segi kehilangan paket, kelewatan dipermulaan, dan kelewatan. Tesis ini menerangkan protocol penghantaran aliran audio yang melaksakan sistem tersebut, susunan sistem, dan bagaimana system tersebut mengenalpasti dan memajukan isu-isu yang lalu. Rekabentuk cadangan telah digariskan untuk menyediakan pengguna/pemberi yang boleh diubahsuai untuk aliran audio menggunakan Real-Time Transport Protocol (RTP) yang melibatkan rekabentuk berasaskan Java Media Framework API. Protocol RTP adalah piawaian protocol internet untuk penghantaran data secara masa-nyata, termasuklah audio, video, dan boleh dilaksanakan dengan menggunakan Java Media Framework (JMF). Java Media Framework library dan Real-time Transport Protocol (RTP) untuk penghantaran audio digunakan sebagai alat pembangunan.

Sistem yang telah dibangun dan direka di dalam tesis ini, bersama keputusan ujikaji membuktikan bahawa system tersebut boleh dilaksanakan dengan jayanya. Model percubaan system ini telah dilaksanakan dan diuji di dalam Makmal LAN dan LAN UPM untuk menganalisa isu-isu yang telah di nyatakan sebelum ini.

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I certify that an Examination Committee met on **23<sup>th</sup> March 2004** to conduct the final examination of **Ibrahim Asaad Aref** on his **Master of Science** thesis entitled “**Audio Streaming System Using Real-Time Transport Protocol (RTP) Based On Java Media Framework (JMF) Application Programming Interface (API)**“ 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:

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## **DECLARATION**

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

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