PERFORMANCE ANALYSIS OF VIDEO CODEC (H.263+, H.264) FOR VIDEOCONFERENCING OVER WIRELESS LOCAL AREA NETWORK (WLAN)

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ABSTRACT

Videoconferencing is widely use all over the world, whether on business corporate, distance learning, or as in video call. Market high demand on good video quality helps corporation in reducing financial needs on travelling, but keep the budget-in for bandwidth usage. High bandwidth or good compression techniques. Both maintain a high video quality. Videoconferencing have high sampling rate, to convert audio and video analog to digital signals. Therefore, there is need of high bandwidth to support the sampling rate. Inadequate bandwidth may lead to pixilation, where, in congested network, a sample can be received in out of sequences. Thus, undesirable video-image quality.

In this research, H.263+ and H.264 will be use as video engine. A real device simulation is used to demonstrate the selection of video codecs with good quality of video resolution. Testbed are measure based on video resolution (240p, 480p, and 720p). The test is carry out in predefine wireless network (WLAN) whereby, performances are measure on MOS score, packet jitter and packet loss.

Convergence of applications (file sharing, video steaming and etc.) in internet put fluctuation in the network. Therefore, simulations are tested in optimum network (utilizing the bandwidth without any disturbance) and in 'converge' network (network with other traffic) to observe the behavior of each codec in different resolutions. A codec with high quality of video resolution is expected to perform in the simulations.

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INTRODUCTION

1.0 Introduction

Mainstream of videos are increase in directly proportional over year. Corporates have been using both audio/voice and video communication. Recent advancements in communication technology have cut lose some unnecessary budget in operational values of businesses. VoIP that enables users to make calls over the Internet and Video over IP that actually, enabling users to meet and see other user. This research will study on the functional of Video over IP across global stream and hardware devices.

Developments of Video over IP have reached many sectors of communications such as Video on Demand (VOD), Audio and Video on Demand (AVOD), digitized video, video streaming, interactive video, and real-time audio/video. In presenting videos over the Internet, video signals must be captured and digitalized before it can be stream. Therefore, devices are needed to capture sequence of motions (analog signal) and turning it into digital signal that can be transmitted over the Internet.

From developments of Video over IP, video presentations have been grouped that is video broadcasting, Video on Demand (VOD) and videoconferencing. Video broadcasting and Video on Demand are categorized as one-way transmissions, while video conferencing is a full duplex. Therefore, this research will specify on videoconferencing that provide real-time communication.

Videoconferencing allows people in different location to communicate (see and hear) as they faced and converse each other in digitalized world. Videoconferencing is a combination of a full duplex audio and video transmission. It is real time and a two-way communication. Videoconferencing can be point-to-point (one-to-one user communication) or multipoint (multiple users in communication).

Performance and quality of videoconferencing can be check using monitoring tools/network analyzer. Monitoring tools can assessed the voice quality of degraded calls, troubleshooting error and network problem, and meaning of logged event. Monitoring tools also provided a performance checker that will check the video quality and audio quality based on scales (voice quality measurement) –whether MOS or R Factor.

1.1 Problem Statement

Performance of videoconferencing are not only determined by the equipment, bandwidth usage but also by the type of codecs (video and audio) used. Some conferencing clients offer the selection of codecs to user, whereas user can choose the quality of its own video and speech sessions in basis of high quality codec (higher bandwidth requirement or lower quality codec (lower bandwidth requirement).

With the various available video codec in market, mainly H.263+ and H.264 plus the availability of video resolution ranging from 240p up to 1080p, it is a challenge to identify which video codec and resolution perform the best at a given wireless network condition. Many studies have been done with focusing on video codecs without taking the resolution into consideration. Furthermore, previous studies also did not focus on effect of network bandwidth toward video conferencing.

The question here is, how the performance of a specific video codec and resolution behave when being tested on a ubiquitous campus wireless network. With the variety of Internet applications, network congestion cannot be avoided. Therefore, network bandwidth fluctuated and affects the quality of videoconferencing.

1.2 Objective

The primary objectives that need to be achieved in this study are:

i. To simulate videoconferencing session using H.263+ and H.264 video codec with resolution of 240p (low resolution), 480p (standard resolution), and 720p (high resolution) on predefined wireless LAN network.

ii. To analyze the performance of H.263+ and H.264 video codec with resolution of 240p (low resolution), 480p (standard resolution), and 720p (high resolution) on predefine wireless LAN network based on scales in the MOS, R-Factor, packet loss, packet jitter, average bandwidth uses and total traffic.

iii. To suggest the best video codec and resolution based on MOS, R-Factor, packet loss, packet jitter and the resolution quality for the predefined wireless network.

1.3 Scope

Limitations during this research are focused on the following:

i. Video codecs use in the simulation is H.263+ and H.264
ii. Speech codec use in the simulation is Speex.
iii. Resolutions to be use in the simulation are 240p, 480p, 720p. These resolutions are grouped into three categories: 240p of low resolution (128 kbps), 480p of standard resolution (256 kbps) and 720p of high resolution (512 kbps).
iv. Implementation of wireless local area network is set on IEEE 802.11n standard in wireless local area network.
v. Simulation is based on SIP architecture only.
vi. Performances for conferencing are each measures and scales using MOS (Mean Opinion Score), R-Factor, packet loss, packet jitter, average bandwidth uses, total traffic and the resolution quality.

1.4 Thesis Organization

The research consists of five chapters:

Chapters 1 provide the overall overview of the thesis. Here, the problem statement will be introduced. Then based on the problem statement, the objective of the research is being defined. Lastly, chapter one also will explain about the research scope.

Chapter 2 introduces video codecs that will be used in this research project. Protocols and resolutions type or categories involved in the video conferencing will be explained in briefly. The literature review is organized in a way that readers can understand this.

Chapter 3 explains the methodology that will be used to progress in this research. Step by step process or phases will be elaborate in this research. All tools used in this research will first introduce in this chapter.

Chapters 4 design the model or know as architecture that will be developed in order to perform the test. It then followed with the continuously design on data analysis.

Chapter 5 explains the result and discussion. One by one analysis and result on parameters will be discussed. Based on this chapter, summary of selection will be made on next Chapter.

Chapter 6 concludes all the chapters and the recommendations for future researchers explain most of the configurations of hardware and software involved in the research. Detail test result will be included in this chapter.

Chapter 7 contains all references that were used in this research.

Chapter 2

LITERATURE REVIEW

2.0 Introduction to VoIP, SIP and Video Conference

Nowadays, videoconferencing solution has emerged based on IP protocol – SIP. SIP has long been implementing in many network real-time applications, mainly in VoIP. VoIP is a transmission of voice data over the Internet. Most VoIP applications using SIP rather than H.323 because of its simplicity and easy-to-use standard. VoIP managed to cut out many unnecessary costs for users. With Internet implementation, users can call and receive data for as long as they like within a small budget (pay to the provider/ISP).

Advancements in communication technology and the success implement on VoIP have giving opportunity for video/voice application over Internet (Videoconferencing). Believing that video conference will widely use, this application can be real success to VoIP. SIP also can be used as signaling protocol in video conference. Many have chosen SIP over H.323 in videoconferencing after seeing the features success in VoIP. In terms of voice sent-over, quality will be the same as in VoIP but, for video quality will be differ as no other applications that can be put as a benchmark success.

2.1 Session Initiation Protocol (SIP)

Session Initiation Protocol is an application-layer control that covers on signaling. This is the IETF's standard proposed for establishing VoIP connections. SIP is not responsible for transmitting data; rather its purpose is initiating (create), modifying (coordinate) and terminating (tear down) sessions. In terms of traditional telephone, SIP is the ringing of a phone, the busy tone and the ending of a call. Architecture of SIP is similar to the HTTP, where both are client-server protocol.

SIP depends on SDP[] (Session Description Protocol) to carry out negotiation for codec identification. It is important in a videoconferencing because participants can join and leave dynamically. SDP specifies details such as the media encoding, protocol port numbers, and multicast address. SIP invitations used to create sessions carry session descriptions that allow participants to agree on a set of compatible media types. SIP makes use of elements called proxy servers to help route requests to the user's current location, authenticate and authorize users for services, implement provider call-routing policies, and provide features to users. SIP also provides a registration function that allows users to upload their current locations for use by proxy servers. SIP runs on top of several different transport protocols. SIP services that been provided include: User Location, Call Setup, User Availability, User Capabilities and Call Handling.

SIP system consists of two components: user agents and network servers. User Agents is SIP users' end station that acts as User Agent Client (issue request) and User Agent Server (received and response to requests). Network Servers receives updates on users' current locations, received and forward request to the server that have more information of called party, and determine next-hop server and return address to the client.

SIP messages are used for client-server communication. Figure 1 shows the basic SIP operation using SIP messages communication. Method in SIP messages:

Method (SIP Message)	Description
INVITE	Invite user to call
BYE	Terminate connection between two endpoints
ACK	Exchange of invitation messages
OPTIONS	Get information on call's capabilities
REGISTER	Gives information about user's location to SIP registration server
CANCEL	Terminate search for a user

Table 1 SIP Messages

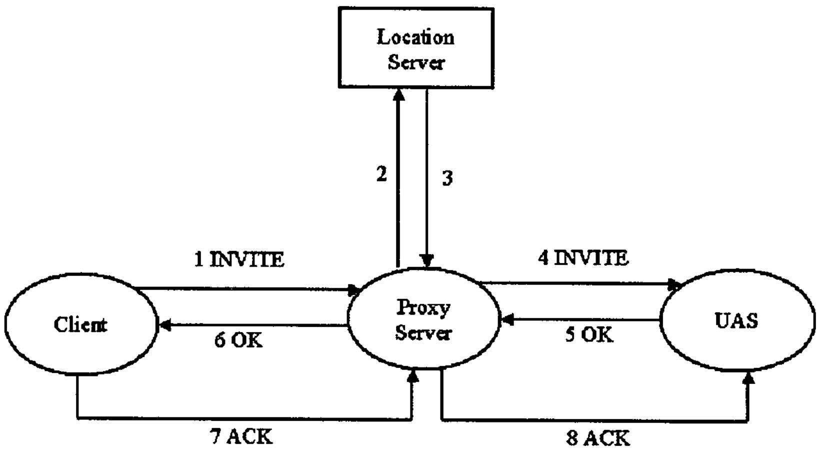

Figure 1 Examples of SIP Operation

2.1.1 Real-Time Protocol (RTP)

Real-time protocols are used by H.323 and SIP as transmission protocol. RTP [9] supports the transfer of real-time audio and video over packet-switched network. RTP protocol is standardized by the IETF, in RFC 3550. RTP is a complex protocol and it is used together with many protocols. Functions that include in RTP: Sequencing, Payload Identification, Frame Indication, Source Identification, and Intramedia Synchronization.

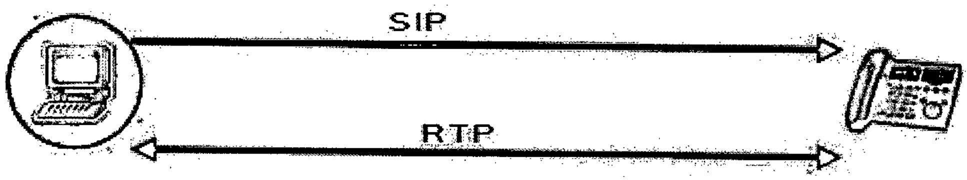

Figure 2 A call using SIP for signaling and RTP for transmission of video and voice

2.2 Video Conference Codec

Videoconferencing depends on video codecs to compress and decompress data being transmitted. Using raw data would increase network resources. Compression and decompression allow limiting network bandwidth. Performance of videoconferencing is related to the video codec's ability coping with different network conditions. The primary codecs used in videoconferencing are H.263 and H.264.

2.2.1 H.263+

The ITU Recommendation H.263+ is a video codec compression designed as a low-bitrate compression format for videoconferencing over narrowband channels. As advancement of ISO H.261, H.263 was used for development of MPEG (high data rates). Generally, H.263 has better quality than H.261. In any circumstances, H.263 has a strong compression component, enabling high performance on movies where there is a little change in frames. H.263+ streaming are packetized for transportation via Real Time Protocol (RTP) over networks. Coding algorithm is similarly used in H.261.

2.2.2 H.264

ITU-T Recommendation H.264 also known as ITU-T H.264, MPEG-4 Part 10, AVC (Advanced Video Coding). It is widely recognized as future video compression for applications such as HDTV services. H.264 was developed to provide high-quality video at lower bit rate than standard MPEG-4 or JPEG. Subsequently, represents a significant benefit to network camera (CCTV, etc.) operations with reduced bandwidth and better pixels.

2.3 Video Resolution

Resolution is important to clearly see an image. Higher resolution means that high number of pixels is used in creating crisper, cleaner image. Video resolution is composed of analog and digital. Analog video resolution is derived from television industry where the image consists of lines, while digital video resolution is derived from digitalized system where an image is made up of pixels. Digital video resolution makes up single image or frame. Video will have many frame or sequential of images to produce moving picture. This number of frames in video is called frame rate. Frame rate show a single second of movement in certain number of frames. Example: QCIF-NTSC has a resolution of 176 x 120 with a frame rate of 30fps, frame per