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An Approach to Develop a new Software (FACEUP) for Video Conferencing Call Using Internet Technology

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Abstract

Multimedia applications over wireless networks are becoming attractive, which are often bandwidth limited and noisy. Video transmission over wireless networks has received much attention recently for its restricted bandwidth and high bit-error rate. Real-time video transmission which requires high bandwidth, low probability of channel error and short transmission delay is the most attractive technology for many applications, such as video-telephony and video surveillance system. Call Admission Control (CAC) is the scheme which regulates multimedia traffic volume in wireless mobile networks and in VoIP (also known as Internet telephony). Most CAC schemes work by regulating the total utilized bandwidth, the total number of traffic calls or the total number of packets or data bits passing a specific point per unit time. If a defined limit is reached or exceeded, a new call may be prohibited from entering the network until at least one current call terminates. This means that the quality of individual calls can deteriorate to a certain extent before new calls are denied entry. Using the basic concepts of CAC, a software named “FACEUP” is developed which can be used to deliver instant messages in the form of text, audio and video. This “FACEUP” software also maintains the good quality video.

1. Introduction

The given input sequence of video is separated into frames. The compression scheme for an input video sequence is based on wavelet coding. The coder achieves good tradeoff between the compression ratio and the quality of the reconstructed video. The compression standards like JPEG and MPEG are based on DCT. First, we estimate the number of bits available for encoding the motion residuals. Second, we select a frame quantization parameter based on motion compensated residuals of the current frame and the previous frame. The aim of the encoder is to take video frames and make an as possible bit stream of them in order to be able to send the video over the internet or a network. The quality needs to be as good as possible and the bit stream need to be as short as possible.

In this paper, we have designed an efficient Software named “FACEUP” and implemented it in video conferencing. FACEUP is a purely JAVA based software which helps to reuse most of the same code over the various operating systems like Windows: 98, 98SE, ME, 2000, XP, vista, 7 Linux: Red hat, fedora, Ubuntu, cent OS, mint and many more. MAC OS X: should work. Java classes can be easily downloaded dynamically over the network and easily integrated with the running application. The fact that FACEUP properly handles both IPv4 and IPv6, especially interesting for direct PC-to-PC communication or in a LAN connection. Encrypted username and password makes it highly secure and reliable. Bandwidth usage is very low due to the various compression techniques used. Text chat is instantaneous (superfast exchange of messages). Very very fast, smooth program. No cumbersome installation/setup required, just copy/download it and run from there.

This software has no complex command-line argument passing, parameter overloading nor any manifest or compilation unit like many other programs. This places our program above them when it comes to user-friendliness. One need not to be any expert programmer to use it, any novice can use our software. To start the program simply double click on the file and it runs like magic.

2. System Requirements

2.1 Software and Hardware device

Software	**Operating System**	**Windows, Linux, Unix, Mac OS X**
	Runtime Environment	JRE 1.6 or higher
	Database/Back end	MySQL
	Browser	IE, Mozilla firefox, Opera, Google Chrome etc.
	Web Server	As per requirement
Hardware	PROCESSOR	Pentium III or higher
	RAM	256 MB (minimum)
	Disc Space	200 MB (minimum)
	Device	Speaker/headphone, Microphone, Web camera or any equivalent device
	Bandwidth	192KBPS(for audio), 512 KBPS or higher (for audio and video)

2.2 Graphical User Interface (GUI)

GUI Interfaces are developed to perform text message transfer among the clients (according to the Fig. 4, to do calls through audio and video. Preferences can be set by using GUI. A new account can be added using a GUI that has also been developed in this paper. This would help to do operations from multiple accounts.

2.3 RTP

The Real-time Transport Protocol (RTP) defines a standardized packet format for delivering audio and video over IP networks. RTP is used extensively in communication and entertainment systems that involve streaming media such as telephony, video teleconference applications, television services and web-based push-to-talk features. RTP is used in conjunction with the RTP Control Protocol (RTCP). While RTP carries the media streams (e.g audio and video), RTCP is used to monitor transmission statistics and Quality of Service (QoS) and aids synchronization of multiple streams. RTP is originated and received on even port numbers and the associated RTCP communication uses the next higher odd port number.

RTP is designed for end-to-end, real-time, transfer of stream data. The protocol provides facility for jitter compensation and detection out of sequence arrival in data, that are common during transmissions on an IP network. RTP supports data transfer to multiple destinations through IP multicast.

Real-time multimedia streaming applications require timely delivery of information and can tolerate some packet loss to achieve this goal. For example, loss of a packet in audio application may result in loss of a fraction of a second of audio data, which can be made unnoticeable with suitable error concealment algorithms. The Transmission Control Protocol (TCP) although standardized for RTP use, is not normally used in RTP applications because TCP favors reliability over timeliness. Instead the majority of the RTP implementations are built on the User Datagram Protocol (UDP).

2.4 CAC

Real-time communications are sensitive to the latency and packet loss that can occur on congested networks. Call Admission Control (CAC) determines based on available network bandwidth whether to allow real-time communications sessions such as voice or video calls and data calls to be established. CAC controls real-time traffic for voice, video and data calls. The CAC offers four main attributes:

- It is simple to deploy and manage without requiring additional equipments, such as specially configured routers.
- It addresses critical unified communications use cases, such as roaming users and multiple points of presence. CAC policies are enforced according to where the end point is located not where the user is homed.
- In addition to voice calls and data calls, it can be applied to other traffic such as video calls and audio/video conferencing sessions.

- It provides the flexibility to enable representation of various kinds of network topologies.

If a new voice or video session exceeds the bandwidth limits of a WAN link, the session is either blocked or (for phone calls only) rerouted to the PSTN.
Administrators define CAC policies, which are enforced by the Bandwidth Policy Service that is installed with every Front End pool. CAC settings are automatically propagated to all Sync Server Front End Servers in a local network.

3. Process of Working

FACEUP is mainly developed to perform the following operations after establishing the connection based on the previously described protocols.

3.1 Client Side Setup

At first, we copy or download the file named Client into our computer. Then we open the file, double-click on the one named Client.jar (or only Client). A login screen (box) will appear.

Fig. 1: Start Up.

Fig. 2: Sign Up.

3.1.1 New User Sign Up

For someone who doesn't yet have an account can create one by clicking on the "NEW USER" button and fill in the details.

3.1.2 User Login

In the "USER NAME" field we type our username and in the "PASSWORD" field we type our password and then we click LOGIN.

o

Fig. 3: Login

Fig. 4: Friend List.

3.1.3 Select your Friend:

After a successful login, the online users will be visible and from there we can select whom we want to chat with.

3.1.4 Communication through Text

FACEUP is used as a tool to communicate over internet through text messages. It mainly refers to direct one to one chat or text-based group chat.

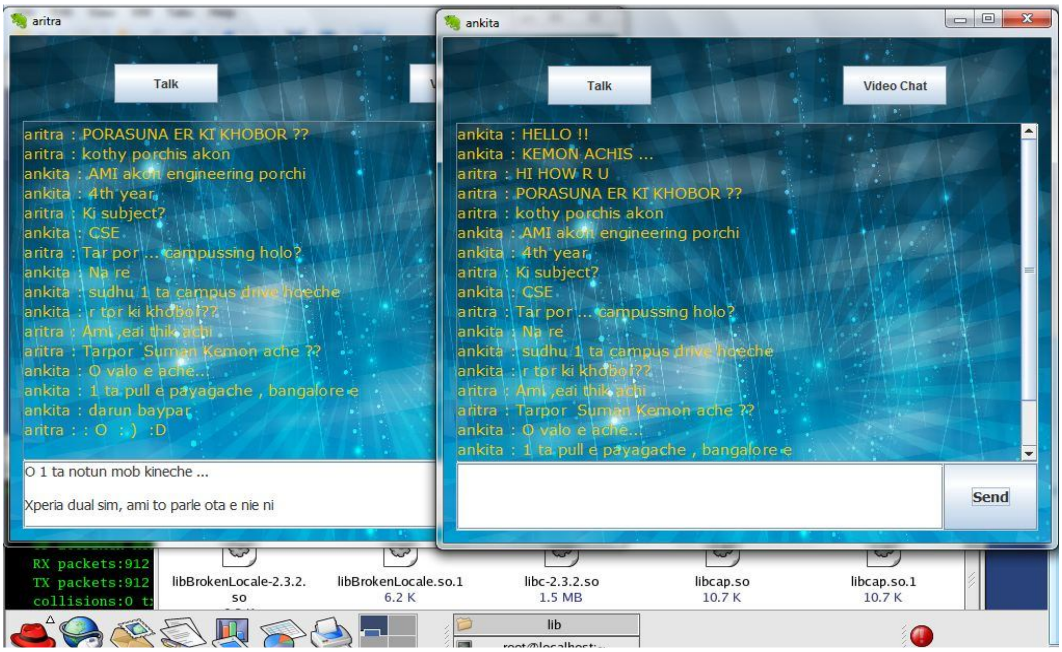

Fig. 5: Text Chat between two friends.

3.1.5 Communication through Audio

In FACEUP, direct audio connection can be established between users. Headphone with microphone is required to perform this operation. Audio system can be chosen but default is Java Audio. The audio device and/or driver must support 8kHz, 16bits audio format.

Encoding scheme used: Raw byte level zip encoding with best compression and default strategy.

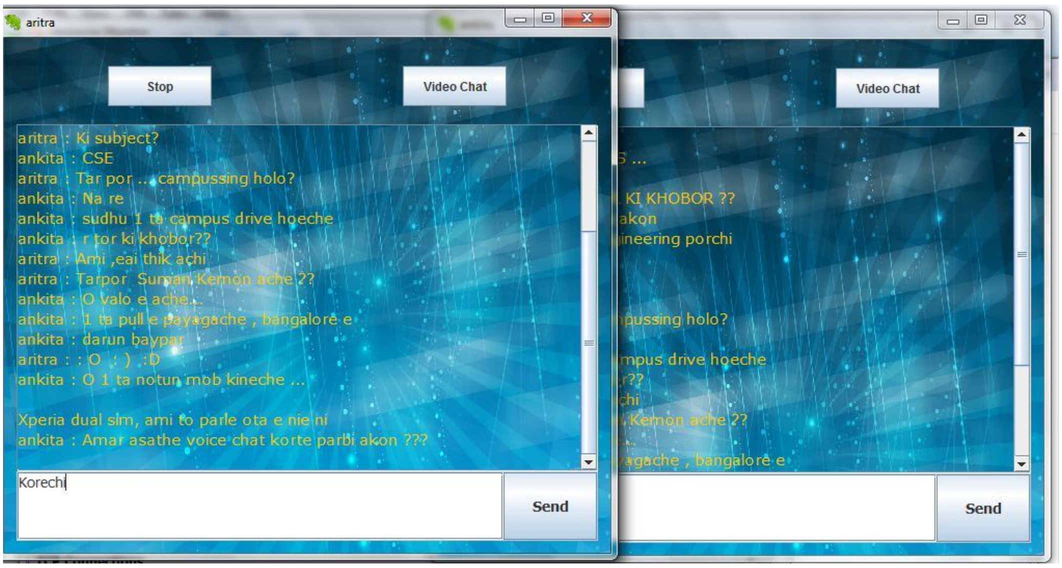

Fig. 6: Voice Chat between two friends.

3.1.6 Communication through Video

In FACEUP, users can communicate through video. For this service, web camera is to be connected to the computer. Once the connection is established between two users, this operation can be performed if one of the users accepts the request of the other's video call. **Encoding scheme used**: H.264 (for video) and mp3 (for audio) encoded in sync.

3.2 Server Side Setup

At the server end connection establishment, database fetching and insertion and communication monitoring etc. are done.

3.2.1 The Server Software:

This is the main server program running at the server side of the organization. At first, double-click on the Server file and then click Start to start the server, Stop to stop it.

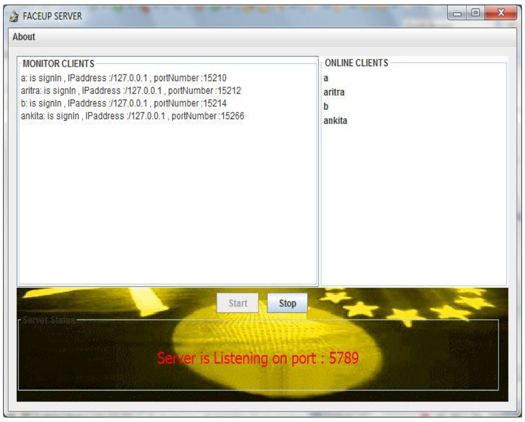

Fig. 9: Communication through Server.

Fig. 7: Video Chat.

3.2.2 The MySQL database:

The databases for user information and user login-password are stored here. We have used MySQL, but other database software can also be used with only a few modifications.

3.2.3 Bye-Bye

If you want to leave the computer for a while, we can easily logout. Just with a single click or even better a shortcut key CTRL+D will do the trick.

4. Conclusion and Future Works

Our software can be used widely and on every aspect whenever there is a need to communicate through computer. Giving and receiving lectures/presentations from remotely located sites from around the world. Conducting meetings between researchers groups and academicians at different universities. This software can be used to deliver instant messages in the form of text, audio and video in an efficient manner. It is useful to deliver online project guidance to the students. It can be widely used in Distance Learning or Job Interviews.

Volume level manipulation and mute facility to be implemented will be future work. Choice of compression and codec can be integrated. Choice for webcam device is also can be implemented, though not a necessity. Some more improvements can be made according to the user/customer demands.

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