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RESEARCH ARTICLE

REAL TIME MEDICAL IMAGE/VIDEO STREAMING

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ABSTRACT

In the medical field the term ‘telemedicine’ getting attention from researchers around the world. This paper introduces system which focuses on real time image/video streaming of medical reports with the help of internet. Microsoft window’s WM encoder is used for data compression and transmission. Data from server machine can be transported easily to the client machine with the help of UDP/TCP protocol. In this paper it is tried to introduce features so that a streaming solution for medical reports canbe designed which will be able to function satisfactorily across all network speeds starting fromas low as 128 kbps. Experimental results show that the image is clear in normal network conditions, and the delayis less than three second. High performances of definition and real-time of the DSA video are both achieved in this scheme.

INTRODUCTION

Telemedicine is the use of telecommunication and information technologies in order to provideclinical health care at a distance. It helps to eliminate distance boundaries and can improve access to critical medical services that would often not be consistently available in distant rural communities using Internet. It is also used to save lives in critical care and emergency solutions. These technologies present communication between patient and medical staff with both low maintenance and fidelity, as well as transmission of medical images and health informatics data from one site to another. Early forms of telemedicine achieved with telephone and radio have been supplemented with video telephony and Information Technologies which give rise to advanced medical diagnostics method [AMDM] supported by distributed client-server applications. Telemedicine application such as introduced above -“Real time medical image/video streaming” can be beneficial to patients living in isolated communities and remote regions who can receive care from doctor or specialists far away without travelling issues.

The challenges of telemedicine include cost of telecommunication and data management who will employ it. Telemedicine can be divided into following three categories

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1) Store and forward services
2) Remote monitoring services
3) Real time services(Interactive service)

In this paper we are focusing on a real time service (Interactive service).

System Overview

This system framework is based on client-server architecture model. There will be a central server with fixed IP, which will accept incoming connections and the clients will be able to establish connections and access to this server. This architecture allows client to be mobile and still able to connect the server. Transport control protocol (TCP) is used to communicate between the client and server. This ensures reliable data delivery which is vital in medical information transfer.

Hardware and Software Setup

The server can be run on windows PC and no additional hardware is needed for server setup. For the client side PC with Windows Media Player only and with NO additional webcam and microphone are required. However, to allow functionalities supported through innovative techniques Microsoft’s WM Encoder corona edition is used at the server side and client side as a ‘codec’ of encoder and decoder of stream. Among the many software platforms considered for the development of this framework Microsoft’s Win32 API

(Windows 32 Application programming interface) as a development platform was found to be most flexible in terms of its ability to manipulate video data as well as the capability to accept multiple devices.

The widespread usage of windows based PC's was also positive factor in choosing this platform .Thus, the framework is developed using Microsoft Visual Studio. Both the server and the client programs are written using C, C++ using COM (Component object model).

Interactive Versus Non-Interactive Applications

Interactive applications such as live image streaming which is introduced here are characterized by the requirement of timeliness such that if the information is not received within specific time limit such that time is sum of end-to-end streaming process which includes-capture encode, stream, transmission, receive, decode, display. Implied by determination of usefulness of information.

Non-interactive applications have looser latency constraints, for example many seconds or potentially even minutes. Examples of non-interactive applications include multicast of popular events or multicast of a lecture; these applications require timely delivery, but have a much looser latency constraint. Note that interactive applications require real-time encoding, and non-interactive applications may also require real-time encoding, however the end-to-end latency for non-interactive applications is much looser, and this has a dramatic effect on the design of video communication systems.

Non interactive applications as opposed to interactive systems mentioned above does not have strict time constraints so that even if information is arrives after delay of some pre-calculated seconds or even minutes it doesn't affect the usefulness of total information. Live multicast of cricket match on several streaming channels with potentially different and some extra delay doesn't affect the information particularly. Procedure within both type of applications contains same processes –capture, encode, transmit, receive, display but with looser time constraints.

Media delivery and control protocols

Understanding the success and lossless transmission of TCP, based on the fact that there are indeed a number of important advantages of using TCP. First, rate control feature of TCP has proven stability and scalability. Second, TCP effectively eliminates the much dreaded packet losses resulting in guaranteed delivery. The International Engineering Task Force has provided some standard protocols for media delivery, control, and description over the world wide Network. The two protocols Real-time Transport Protocol (RTP) and Real-time Control Protocol (RTCP) are major two pillars to support streaming media especially within the domain of interactive application mentioned above. Instead of providing real time services directly they provide the functionality which in turn are used to provide real time service. RTP does not focus on network quality parameters which are described as Quality of service (Ref. Computer Network by Andrew Tannenbaum), instead it focus on tight time constraint which are indeed important for interactive applications which supports the above conclusion.

So what about Quality of service (QoS), here Real-time Control Protocol (RTCP) comes in picture by providing services such as acknowledgements of lost packet, inter arrival jitter, delay, bandwidth utilization and bandwidth, along with inter-stream synchronization and round-trip time measurement. Media Control is provided by Real- Time Streaming Protocol (RTSP).

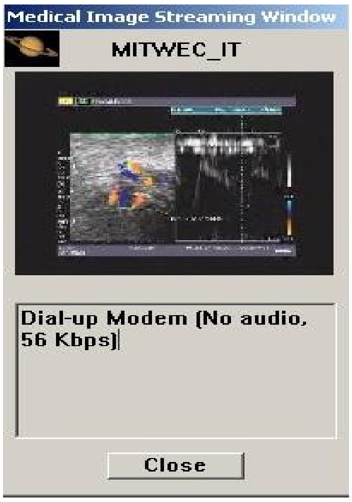
The Real Time Streaming Protocol is used for establishing and controlling media sessions between end points. Like HTTP, RTSP uses Transfer control protocol to maintain an end-to-end connection and, while mostly control messages of RTSP are sent to the server from the client , but some commands do travel from server to client.

CODEC

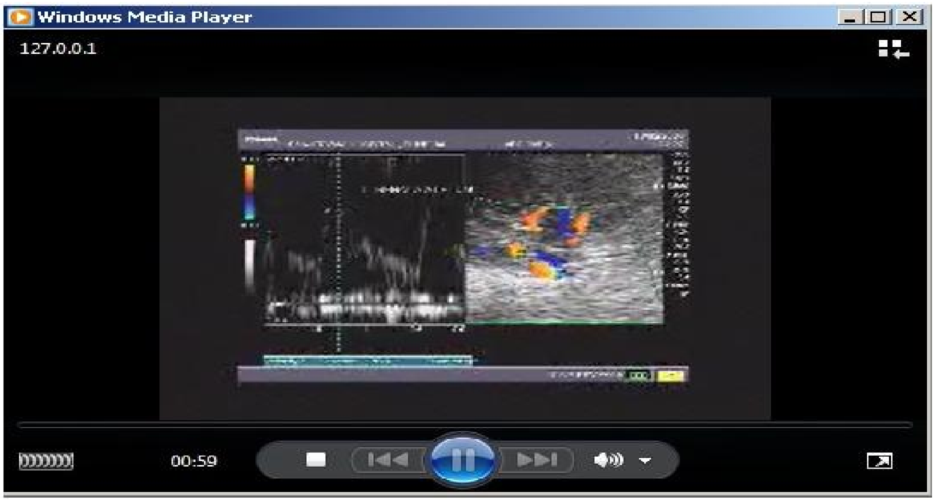
A codec is an algorithm, which can be thought of as a list of instructions that identifies the method used to compress data into fewer bytes. By following these instructions, applications such as encoders and players can reliably compress and decompress data. In the case of digital media content, codecs are used to decrease the content's file size and bit rate. With smaller file sizes and lower bit rates, digital media content can be stored and streamed over a network more quickly. An algorithm used to compress data into fewer bytes is known as codec. So the codec is combination of coder and decoder which is responsible for converting audio or video into streaming bytes. It carries the major responsibility to determine the ultimate quality of streamed data. It can also decrease size of file which contains streaming data along with providing the functionalities to control the bitrate. So the resulted file which is to be streamed is altered such as it can be streamed very quickly to help satisfying rigid time constraints. Windows Media Audio and Video codecs support both VBR and constant bit rate (CBR) encoding. VBR encoding is developed for use when you want to create a downloadable file that has a smaller file size and bit rate without sacrificing sound and video quality.

Microsoft have developed several codec for data streaming over internet. For example Windows Media Audio (WMA) and Window Media Video (WMV). Both support VBR variable bit rate and CBR constant bit rate, in which VBR is specially used to store and forward small files by maintaining their quality. Whereas in CBR the encoder knows the bitrate of the output before the encoding session begins. The same number of bits are used by the encoder to encode each second of data throughout the duration of the file to achieve the target bit rate for a stream. So it is more convenient to use CBR in the presented application. CBR encoding is useful when you want to know the bit rate or approximate duration of a file without parsing the entire file. This is important in live streaming scenarios where the media content needs to be streamed at a predictable bit rate and with consistent bandwidth usage. In CBR we have two options one single pass encoding and the other multi pass encoding.

Single-pass encoding analyses and encodes the data in one single pass with a quick speed which is important for interactive application and is used in constant bitrate encoding. For real-time encoding, Single-pass encoding is usually controlled by the “fixed quality” setting or by the “bitrate range”.



(a) Server



(b) Client

Fig. 1. Screen Shot of Medical Image Streaming

Windows Media Encoder

Microsoft has developed a package of codecs for audio, screen captures, and streaming video with different bitrate options CBR (Constant bitrate), VBR (Variable bitrate), MBR (Maximum bit rate). This package is known as WM Encoder and it provides high degree of configurability for quality and size reduction of streaming data. Microsoft has developed Windows Media Encoder which is used to capture and convert pre-recorded audio, video and live audio, video to Windows Media formats for real time streaming. We can write our own applications in Visual Studio to perform the same functions found in applications.

WINDOWS MEDIA VIDEO CODEC

Windows media video codec compress video or audio into stream of data that is encapsulated in a data container which have extension .WMV. Important thing here is that WMV is not the only format which is generated by WMV codec. In fact the potential fact that WMV codec and WMV format are totally different because it is a type of container which is one of the data container that WMV codec can create such as AVI. WM encoder is one of the software which exports the video in the WMV format. All the videos encoded by WMV codec are playable on Windows Media player. It was built upon Microsoft's implementation of MPEG-4 Part 2.

Comparison of proposed approach and upload and download approach

To illustrate the difference between the proposed systems based on streaming and conventional upload and download methodology. Consider a real time scenario such that there is one sender and receiver. Sender is sending an object to receiver so it uploads that object, means it is given to the third party middleware having cloud storage space (for eg, Google). Suppose this total process requires x time than receiver contacts this middleware and then request to receive this object (here analogous to downloading) and assume this requires x/3 time as most the connections currently used are asymmetric that is downloading speed is greater than upload speed. Total time required is 4x/3. Secondly, unless and until every piece of object is received the object is not useful as the receiver cannot access the object fully.

Now consider the same situation using streaming in which each piece of object is directly transmitted to receiver by sender without any middleware and in sequential manner resulting in eliminating the requirement of storage as well mean time to access (MTTA) the object being usable as soon as the piece of object is received.

So in worst case also, considering the same network conditions in both the situations even though time is dramatically reduced (less than 4x/3).

RESULT AND ANALYSIS

Table 1. Comparison of different methods

Method No.	Description	Time (sec)	Conclusion
1.	Upload and Download (Traditional Approach)	180	Simplified but not applicable for real time
2.	Streaming on Server only (Single Machine)	8	Streaming is implementable & elaborated minimum required time.
3.	Proposed Method	13	Real- time applicable and implementable.

Conclusion

In this scheme, Medical images are streamed. Window Media Video codec technique is used for data's compression and transmission. Data from server to client is sent using TCP protocol. For consistent delivery various media control protocol such as RTSP and RTP are used. Experimental results show that the image is clear across the internet, and the buffer delay is acceptable. Experimental results show that streaming of a medical report via Internet is a feasible scheme.

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