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Multimedia Content Distribution System over Mobile and Fixed Networks

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ABSTRACT

A structure is presented to be able to send audio and video data to a large group of multicasts, based on hyper text browsing by web-based architecture. The investigations show important results; first, acceptable audio and video quality is more usable in weak foundations that are audio range is 5 kb/sec, and video range is 9-100 kb/sec. second, compressing audio and video data is carried out based on standard algorithms. So, receiving resources and playing them by standard decoders are easily done. And third, the structure of transferring audio and video data simultaneously is carried out based on HTTP standards.
KEY WORDS: Multimedia; distribution system; Networks; meta data; HTTP standards.

INTRODUCTION

1. Simultaneous transfer and live streaming multimedia

The aim of this effective structure is providing the field of working with different foundations, availability of different devices for connecting internet (PC, MAC, PDA, PSP, Mobile, etc), and the possibility of utilizing some cases such as: simultaneous and live transfer of digital TVs, video conferences, telemedicine, telemanipulation, remote video surveillance.

Similar software existing in market is not suitable to meet our needs. For example, Microsoft Windows Media set[1] (Encoder, Streaming Server, and Player) is a proper and powerful device which transfers lives and saved audio and video flows. But unfortunately, this set is designed only for the purpose of entertainment. So, high delay is happened during sending data, which is opposite of this system goals. Also, there are some limitations in choosing video codecs. In fact, this is applicable for Windows Media Codecs [1].

Sending data will face long delay in each solutions existing in weak foundations of audio and video like Helix Streaming Server[2] or Darwin Streaming Server[3]. These foundations are able to create some connections. But they have long delay in sending data, and they utilize private protocols and codec in commercial foundations, as well. So, they are not pervasive in any case [2].

Videolan[4]system is an open source system, and unlike the above mentioned devices, they are carried out for the purpose of researches, not for economical goals. This system possesses high transfer ability and more flexibility in comparison with commercial cases. But the weak point is that live audio and video transfer is just possible via MPEG TS (Trans-port Stream). Regarding to use of MPEG-TS/UDP/IP storage memory, this system occupies twice as much as Buffer than mentioned memory. So, during using this system, limitations in application will be revealed in existing networks, and they do not offer a good performance.

Tendency to video data sending has been increased in the recent 10 years. A paper about these activities process was published in 2000 by Lu [3]. In this paper there are some reports and topics about processes of signals related to video data sending and some solutions are suggested. In fact, a case of video data sending is discussed from different viewpoints. For example, Lu discusses saving on computer resources used by a service provider [4]. Some other activities attribute video data sending and current [5] and [6] to system structure. In [7] activity, Conklin talks about some communicational methods for video data sending via high bandwidth internet. In [8] activity, Guo has investigated live video data sending by transfer devices, and has explained about a structure for simultaneously files receiving. It should be mentioned that this algorithm is related to receiving files, not living resources.

[1] www.microsoft.com/windows/windowsmedia
[2] www.realnetworks.com/products/
[3] Developer.apple.com/Darwin/projects/streaming/
[4] www.videolan.com

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Nowadays we can call many cases such as VDOLive[5], Streamworks[6], Vosaic[7], VivoActive[8], InterVU[9], RealNetwork[10] in transferring and broadcasting audio and video simultaneously. VDOLive, Stream work, Vosaic and RealNetwork are usable based on exclusive systems of service provider under communication protocols UDP/IP and RTSP. This protocol is one of the unreliable protocols for lost and undelivered data. This protocol is closed by firewalls in most of networks, because of insecurity. StreamWorks, Vosaic and interVu, as well, are used as MPEG based on compression, and they need high bandwidth for sending data. Generally, most of these solutions have been designed for the purpose of being used in LAN networks with high capacity[9].

Some other networks are suggested for sending audio and video data via network. They are usually designed for high capacity and long bandwidth foundations, because of utilizing MPEG-4 or exclusive protocols[10] such as SCTP, MMS, and RTSP, or at least they need exclusive equipment and decoding software[11].

The goal, here, is to transfer and broadcast audio and video data, included weak-foundations networks, low security capacity, promotion of meta data.

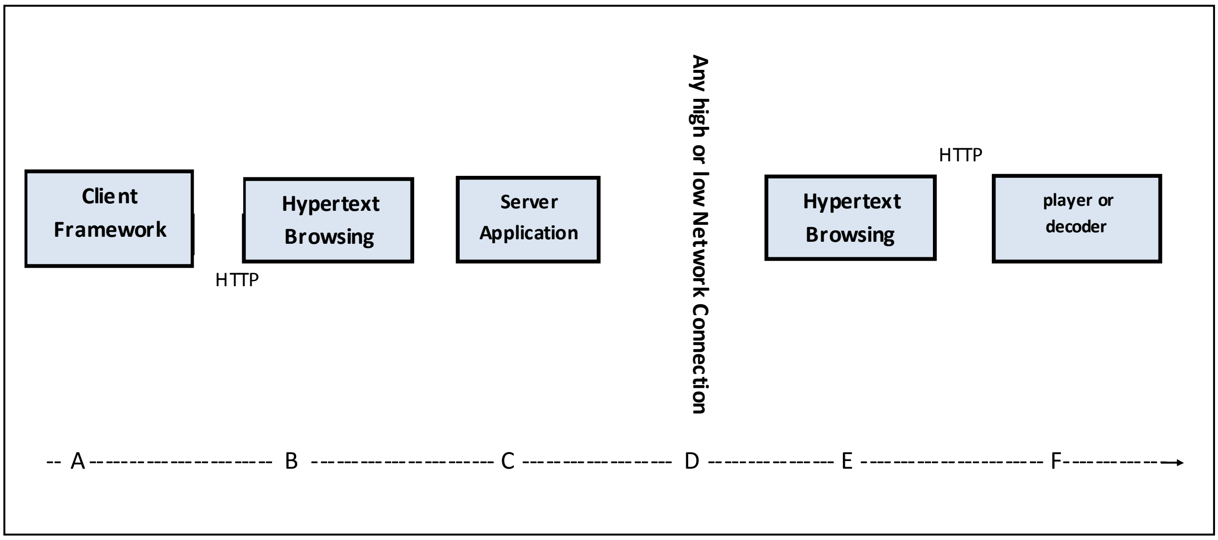

Fig. 1

Some characteristics have made this system proper in order to transfer live audio and video data via long bandwidth and wireless networks. For example, an advanced system for editing problems between receiver and service provider, exclusive performance of Buffering in a harmonic technique between sent and delivered data.

Needed foundation for a transferring meta data system is http standard protocol in data process structure; receiver-service provider- final user. In this part we explain about "receiver- service provider" architecture, with details related to compression algorithms. System synchronization technique will be explained, as well.

In order to accomplish the mentioned goals in transferring meta data, design has been done in a way that no special software will be needed. In fact, transferring and broadcasting live data will be applicable with HTTP standard web server; some independent devices of operating system are used in order to be compatible with different kinds of plat forms, and library functions related to HTTP protocols are utilized, as well.

Saved or live resource of audio and video data is received by receiving meta data hardware on the network based on hyper text browsing by a framework of web-based architecture. This system is composed of receiving and service provider parts and central algorithm.

Transfer and broadcast Meta data simultaneously is restricted to communication mechanisms of HTTP protocols. HTTP place is in Application layer; so, in this situation, used Transport layer protocol (used by stack TCP/IP)

[5] http://www.vdo.net
[6] www.xingtech.com
[7] www.vosaic.com
[8] www.vivo.com
[9] www.intervu.com
[10] www.realaudio.com

creates a communication in which data are given to communication protocol in a way that there is no need to a contract more than application layer to handle lost data. In this situation transferring meta data will be treated like any web-based object. So, at the beginning, live Meta data broadcasted in network will be transferred like an html page or a JPG video file.

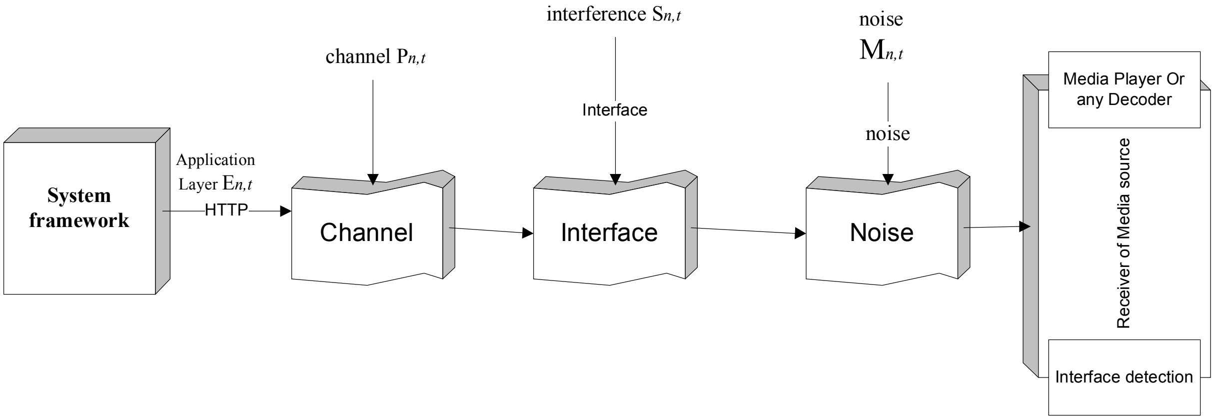

Fig. 2

Saved or live resource of meta data makes the YUV[11] frame ready for reviewing in basic structure of Hypertext Browsing. Moreover, based on Buffer structure in prepared equipment of meta data resources, received data will start and stop system discontinuously based on HTTP transfer. Since access to transfer framework communication is executed independently of protocol and Buffer, this start-and-stop system will optimize analyze process. It should be mentioned that if receiving data volume of final user is more than code loading speed for a short time, any video data will not be lost. This system negative point is that Buffer faces with delay during its operation. So, fewer Buffers are needed to be occupied, and if this is not done, receiving data speed will not be more than code loading speed.

Since one of the system goals is working with network different capacity, it is necessary to investigate network foundation before anything else. So, an algorithm is created for the purpose of a proper meta data sending for codec based on final-user communication.

In Fig 2, sent audio and video codec is shown in which it is proper for the purpose of transferring and receiving meta data simultaneously. In this figure, $E_{n,t}$ is transferred audio and video data in time t on Nth foundation, $P_{n,t}$ is channel answer in time t on Nth foundation, $D_{n,t}$ is received audio and video data in time t on Nth foundation, $S_{n,t}$ is input data in time t on Nth foundation.

When there is a request between final user and center and receiver (that is input data is existed), the received data under system structure will show below equation:

$$D_{n,t} = E_{n,t} P_{n,t} + S_{n,t} + M_{n,t} \qquad (1)$$

Without having input data top equation will be as below:

$$D_{n,t} = E_{n,t} P_{n,t} + M_{n,t} \qquad (2)$$

In codec algorithm, it is assumed that the operation is taken place in fixed or semi fixed channel. So, communication time and bandwidth is fixed, and $P_{n,t}$ could be equal to P_t.

Based on the mentioned issues, desired state in a proper time for a proper foundation will be as below:

$$\sum_{n} D_{n,t} E_{n,t}^{*} = \sum_{n} P_t E_{n,t} E_{n,t}^{*} + \sum_{n} S_{n,t} E_{n,t}^{*} + \sum_{n} m_{n,t} E_{n,t}^{*} = \sum_{n} P_t E_{n,t} E_{n,t}^{*}$$

Video and audio compression standard is based on H.xxx system and G.xxx system, respectively. So, this algorithm makes the video data optimized with high quality in low-bit-rate based fields.

The packs of proper audio and video data are sent based on codec and H.xxx compression standards, and they are changed into UDP diagrams based on web-server based architecture. UDP is processed earlier in network. So, they have priority to TCP; but this characteristic might cause some audio and video data loss.

[11] Leuchtsignal, Farbdifferenzsignale U and V (PAL SECAM TV).

During sending data, a few numbers of packs are damaged or lost. Based on the experiments, the frame work system on web-server-based architecture changes communication behavior of transferring data. This is an efficient system. In receiving part, a structure based on standard decoding of compression standards converts UDP packs to audio and video packs again, and this is done by adapting sent data and delivered data, and the decoder is converted into RGB by final user's software. Even if receiving data speed and frames broadcasting speed is equal, packet produce speed (in direction of transfer framework based on http) and packet review speed (in decoder direction) may be different for some reasons. Moreover, buffer in UDP receiver network plays an important role in decreasing negative effects of mentioned cases. Method of using network buffer should be investigated carefully for two reasons; first, delay of replay is directly related to temporary memory space occupation. Second, if packet produces speed is more than packet review speed in adequate time, buffer might be overfilled and therefore received packets will be lost. Regarding to these reasons, an intensive and compatible structure has been added to the system dynamically, in order to increase or decrease buffer quantity. During the operation, when buffer is too full (80%), this intensive structure makes buffer quantity twice as much as before; and if it is too empty (20%), the structure halves buffer quantity. The threshold level for activating this intensive structure is determined experimentally and depends on different conditions.

The above reports imply that it is better for buffer space to be empty. But sometimes decoder must wait for information receiving. So, it has negative effects on replaying video and even audio data, and increases frames decryption time. So, it is suggested to use a simple algorithm in order to access the best balance level to maintain review level between 10% and 30%.

If this system is used, temporary memory space occupation is close to zero which shows the least delay amount.

2. Conclusion and future goals

In this paper, we explained about a system transferring and broadcasting audio and video data simultaneously, and general availability possibility in existing networks, included networks with weak foundations, low capacity and security which try to promote audio and video data sending, and have the least delay amount.

The goal is simultaneous transferring and broadcasting live audio and video data in order to transfer digital TV data, video conferences, telemedicine, and telemanipulation with delay. However, we investigated the existing solutions and concluded that any of these cases will not meet our goals. So, we used compression standards based on http protocols with interface algorithms and UDP/IP system, and we accomplished our goals.

We investigated transferring and broadcasting meta data with a system of audio and video data ready to be sent based on hyper text browsing, in order to reach balance between minimum delay and maximum possible quality in video data sending. The attempts in this field are the symbol of offered solutions effectiveness.

We intend to develop the system based on other compression systems and decryption ones; one of the other future goals is to investigate the possibility of performing offered method on low-calculating-ability systems. Moreover, some activities and long-term experiments are being carried out.

REFERENCES

[1]. http://www.microsoft.com/windows/windowsmedia/, Last accessed: 10 October 2007.

[2]. Andrea Prati and Giovanni Gualdi, Rita Cucchiara. University of Modenaand Reggio Emilia Via Vignolese, 905/b - 41100 Modena, Italy - Low-latency Live Video Streaming over Low-Capacity Networks. 2007 Page 2

[3]. J.Lu. Signal processing for internet video streaming- are- view. In Proc. of Conf on Image and video communications and processing, pages 246 –259 ,2000.

[4]. M.-T.Lu,C.-K.Lin,J. Yao,and H. Chen. Complexity-aware live streaming system.In Proc.of IEEE Int'l Conferenceon ImageProcessing , volume1, pages 193–196, 2005.

[5]. G.Conklin, G.Greenbaum, K.Lillevold, A.Lippman, and Y.Reznik. Video coding for streaming media delivery on the internet.IEEE Transactions on Circuits and Systems for Video Technology, 11(3):269–281 ,Mar.2001.

[6]. M.Guo, M.Ammar, and E.Zegura.V3: a vehicle-to-vehicle live video streaming architecture. In Proc. Of IEEE Intl Confon Pervasive Computing and Communications, pages171–180,2005.

[7]. G.Conklin, G.Greenbaum, K.Lillevold, A.Lippman, and Y.Reznik. Video coding for streaming media delivery on the internet.IEEE Transactions on Circuits and Systems for Video Technology, 11(3):269–281 ,Mar.2001.

[8]. M.Guo, M.Ammar, and E.Zegura.V3: a vehicle-to-vehicle live video streaming architecture. In Proc. Of IEEE Intl Confon Pervasive Computing and Communications, pages171–180,2005.

[9]. Kam-Yiu Lam , Calvin K. Chiu, The design of a wireless real-time visual surveillance system, Multimedia Tools and Applications, v.33 n.2, p.175-199, May 2007 [doi>10.1007/s11042-006-0056-9].

[10]. Andrea Prati and Giovanni Gualdi, Rita Cucchiara. University of Modenaand Reggio Emilia Via Vignolese, 905/b - 41100 Modena, Italy - Low-latency Live Video Streaming over Low-Capacity Networks. 2007
[11]. Quaritsch, Markus, Wiesinger, Mario, Strobl, Bernhard and Rinner, Bernhard (2008), "An Adaptive Multi-Purpose Transmission Scheme for H.264 Encoded Video in Wireless Networks", International Symposium on Communication Systems, Networks and Digital Signal Processing.