MEDIA SERVER FOR VIDEO AND AUDIO EXCHANGE BETWEEN THE U-70 ACCELERATOR COMPLEX CONTROL ROOMS: CURRENT STATUS AND PERSPECTIVES

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Abstract

The dispatching system for audio and video exchange between various U-70 technological subsystems has been developed. The system architecture employs the original approach to the online streaming using the method of progressive download over HTTP. The system can be built on the basis of any one-pass decoding media container. Ogg format with Theora/Vorbis codecs has been chosen as an appropriate container.

The system is designed as a distributed software complex and consists of several executive components: Retransmitter, a number of Media Stream Coders and a number of clients. All the components can run either on separate computers or on the same computer. Media stream Coders consume streams from media sources, turn it into Ogg stream and send it out to Retransmitter. The possible media sources are: IP-cam, web-cam, computer screen along with microphone. A client is able connect to Retransmitter using any web-browser with Ogg Theora/Vorbis support. The range of values 250-350 ms has been achieved for the latency between real action and client video. The developed software is a framework for diverse media exchange systems that could be built on it's basis. Several possible applications are discussed in the scope of the U-70 Accelerator Complex technological needs. This could play an important role in personnel safety system while perform maintenance tasks in an extreme work environments such as the Beam Channel.

INTRODUCTION

The aim of the work is to develop a Media Server system for audio and video exchange between U-70 Accelerator Complex technological subsystems. The requirements for Media Server:

- the client software can run on different operation systems;
- the system software must use open-source free algorithms and libraries;
- do not use any special designed programs (plug-ins) on the client side;
- low latency between actual presentation action and the action coming up to a client screen;
- simultaneous communication between several clients.

The Media Server software employs the original approach to the online streaming. This approach uses the progressive download feature of HTML5 [1] standard for getting a media stream [2] through the tag <video>. Considering particular features of progressive download, only stream-oriented media container should be used. In other words, there must be one pass decoding stream container with no seeking. All in all, Ogg with Theora /Vorbis codecs [3] has been chosen.

The first report on the Media Server was reported at RuPAC 2014 [4]. Since then the software has endured significant changes in ***system structure*** and ***client software***.

To date the Media Server [5] has transformed into distributed dispatch system which consists of two separate executives – the central module called Retransmitter and several programs called Media Coders. Both Retransmitter and Media Coders can run on separate computers or on the same computer, they communicate with each other through Windows socket gear.

THE SYSTEM STRUCTURE STATUS: THE DISTRIBUTED EXECUTIVES

The dispatching system as a whole consists of several components, as follows:

- Media Sources;
- Media Stream Coders;
- Retransmitter;
- a set of web-pages for client.

DBMS SQL is used to control the system workflow, web server allows clients to download the web pages.

The structure of the dispatching system is shown in Fig.1. The scheme depicts the case of three presentations being viewed by four clients. Client 1 is connected to presentation 1, client 2 is watching presentations 1 and 3, client 3 is watching presentation 3, client 4 is watching the archive. Presentation 2 is being watched by no one.

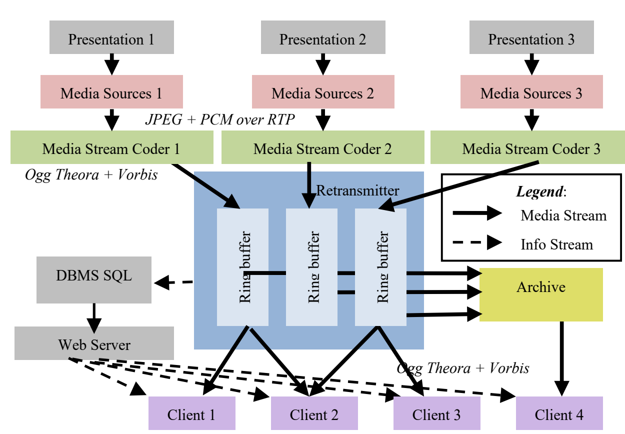

Figure 1: The structure of the dispatching system.

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Retransmitter is going to send online media stream from presentation 2 immediately after any client connects.

Presentation is an action which is being played out and which is need to be transmitted to a client. For example lecture is a presentation.

Media Source is a device for digitizing a Presentation and sending it to a consumer. It can be either a single IP-cam, a web-cam, a microphone, a screen image. But strictly speaking, Media Source is the supporting software for these devices.

Media Stream Coder (MSC) is an executive which receives audio samples and video frames from Media Source, takes them through a codec, then incapsulates it into a media container. The media stream thus obtained is being sended to Retransmitter through the LAN/Internet.

Retransmitter is an executive which receives the media stream from MSC and sends it to clients. Retransmitter is able to get media streams from several MSC and serve several clients simultaneously. One media stream (a stream from one presentation) can be transmitted to several clients. Rettansmitter has a separate circle buffer for each MSC in order to smooth down the roughness in the receiving stream flow from MSC to a client.

The status information on current presentations is mirroring into the data base. Web-pages for available presentations are being created dynamically based on information stored in the data base. Client gets a list of current presentations marked with name and small preview image. After choosing one or several presentations of his interest a client activates some URLs of selected media streams and begins to play back them.

THE CLIENT SOFTWARE STATUS: REDUCING PLAYBACK LATENCY

Playback latency is a difference between the presentation's event time and the time of corresponding event on the client's screen. The playback latency is the sum of three terms - a) delay for media source digitization, b) delay for coding/decodig and transmitting over LAN, and c) delay caused by the frames accumulation into the browser's client buffer.

The first delay depends on media hardware and we are not able to reduce it. To date the IP-cam DLink DCS 3430 is used with 170 ms digitization delay.

The second delay depends on network bandwidth and computer power, the amount is roughly a few dozens of milliseconds.

As to the third delay, modern browsers are able to playback progressively with a several video frames left in the playback buffer. When the frame frequency equals to 20 frames per second, the buffer delay stands at 200 ms. It's a theoretical minimum value but in reality browser software is quite sensitive to changes in transmit rate. If the incoming bitrate reduces, the browser may stop playback spontaneously until the playback buffer increases up to several seconds, after which browser will continue playback. As a result the delay will increase accordingly. To address the last issue a smooth buffer correction algorithm was developed. The algorithm watches constantly the playback buffer and tries to reduce it up to the minimal value.

With the help of smooth buffer correction the following playback latencies have been achieved:

- 350 ms when IP-cam is used as a Media Source (cam digitization accounts for 170 ms of this total);
- 250 ms when computer screen along with audio controller are used as video and audio sources respectively.

PERSPECTIVES OF APPLYING MEDIA SERVER IN U-70 ACCELERATOR COMPLEX TECHNOLOGICAL STRUCTURE

Scheme 1. Live Webcasting of the Meeting

The conference which takes place in the Conference Room (Fig. 2) is broadcasting to Clients 1 and 2. They can watch video from two IP-cams, listen to audio from two cam's microphones and watch the screen of the presentation laptop. Client 3 is watching through an archive.

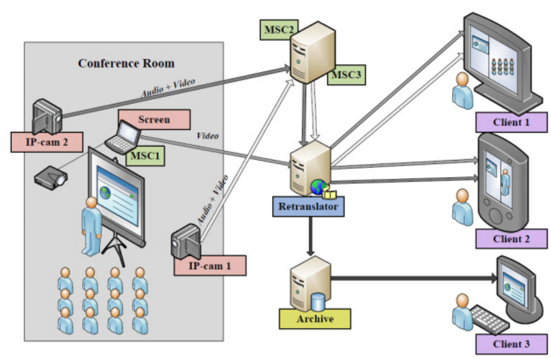

Figure 2: Live webcasting of the conference. The label colors correspond to those specified in Fig.1.

Required hardware and software configuration for the dispatching system's components is:

- A presentation computer on which Media Coder for the screen (MSC1) is running.
- Two IP-cams (IP-cam 1 is pointed towards audience, IP-cam 2 is directed towards speaker). Media Coders (MSC 2 and 3) are running on a special computer. The media coders incapsulate images and audio from the cams into a video stream and send it to Retransmitter.
- Retransmitter can run on the standalone computer or the same computer which MSCs are running on. Retransmitter performs the video+audio streams archiving into a set of media files.
- Clients can watch video from any of the cams, listen to one of the audio. At the same time clients can watch video of the presentation screen.

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Scheme 2. Simultaneous Supervision of Multiple Workers Working in the Extreme Environment

Figure 3 depicts a part of a hazardous area filled with heavy machineries with uneasy access, for example a Beam Channel. Two operators are performing some time-critical maintenance tasks. Both can communicate with the dispatcher located in the Control Room (CR). The communication is organized through the network of WiFi stations. Each operator has an IP-cam with self-contained power supply. The cam sends the video+audio stream to Retransmitter and therefore the dispatcher can see everything the operators do. A feedback with dispatcher is implemented with the help of a mobile phone connected to the Retransmitter via WiFi. Operators can listen to the dispather's audio through the web-page.

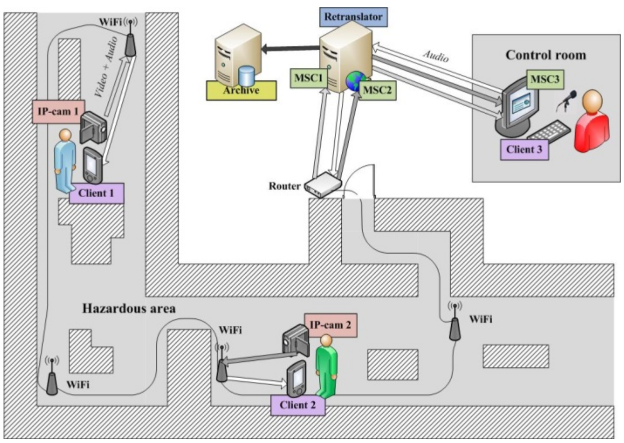

Figure 3: This is how two operators are communicating with the dispatcher and with each other.

Required hardware and software configuration for the dispatching system's components:

- Video+audio sources 1 and 2 are the IP-cams, audio source is a microphone connected to the computer in the Control Room.
- Media Coders 1 and 2 are running on the mutual server, which is shared with Retransmitter. Media Coder 3 is running on the control computer, it makes an audio stream from the dispatcher.
- Both Retransmitter and MSCs are operating on the same server.
- Loud speakers/head phones are connected to the audio card of the control computer in the CR.

CONCLUSIONS

The presented work is dedicated to the implementation of the original live streaming technology in the scope of U-70 Accelerator needs. The technology is based on the progressive downloading method over HTTP for the media stream.

The media stream format has to meet the requirements of a) one-pass encoding/decoding feature (which means that there is no need in getting the follow-up media in order to decode the current one) and b) to be open and free. The Ogg format with Theora/Vorbis coding scheme meet both of above-mentioned requirements.

With using web browsers as a playback unit a problem arises for voluntary internal buffer increasing. This, as a consequence, leads to the increasing of the latency between the live presentation actual time and the playback time in a browser. To address this issue a smooth buffer correction algorithm has been developed.

Main characteristics of the developed dispatching system are:

- a web browser can be used as a client playback unit in order to playback video and audio. HTML5 feature (<video> tag) is used, there is no need in additional plugins and extensions.
- low latency (250 ms) allows dispatching system to be employed in two-way real time communications scheme;
- an arbitrary number of clients can be connected to a single presentation;
- the used media stream format is free and open.

The implementation has the components of two types (MSC and Retransmitter) which communicates with each other through the LAN. The executives can run either on different computers or on the same computer. Any number of components can be organized into a distributed dispatching system for a wide range of objectives. Thus the system is distributed and its practical implementation can be built from a number of software blocks in various combinations. For example:

- video/audio connection between two or more CRs;
- live web casting of the meeting on the Internet;
- simultaneous supervision of multiple workers who are working in the extreme environment.

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