International Journal of Scientific Research in Science and Technology
Print ISSN: 2395-6011 | Online ISSN: 2395-602X (www.ijsrst.com)
doi : https://doi.org/10.32628/IJSRST

Study of Application Sharing Approach of Social Media Networking Sites

Nabajyoti Sarma[1], Dr. J. P. Tripathi[2]
[1]Research Scholar, Computer Applications, B R. A. Bihar University, Muzaffarpur. Bihar, India
[2]Department of Mathematics, R. N. College,Hajipur, Bihar, B R. A. Bihar University, Muzaffarpur. Bihar, India

ARTICLEINFO

Article History:

Accepted: 05 Jan 2023
Published: 30 Jan 2023

Publication Issue
Volume 10, Issue 1
January-February-2023
Page Number
713-718

ABSTRACT

In this paper we present about the application sharing approach of MCS. Multimedia refers to content that uses a combination of different content forms. This contrasts with media that use only rudimentary computer displays such as text-only or traditional forms of printed or hand-produced material. Multimedia includes a combination of text, audio, still images, animation, video, or interactivity content forms. An individual multimedia designer may cover the spectrum throughout their career. An example of a software toolkit that assists in development of shared computer applications is Bellcore's Rendezvous system (language and architecture). Shared applications may be used as conversational props in tele-conferencing situations for collaborative document editing and collaborative software development.

Keywords : Multimedia, Networking, MCS, Transmission line, MBone, CSCW.

I. INTRODUCTION

Sharing applications is recognized as a vital mechanism for supporting group communication activities. Sharing applications means that when a shared application program (e.g. editor) executes any input from a participant, all execution results performed on the shared object (e.g., document text) are distributed among all the participants. Shared objects are displayed, generally, in shared windows.

Application sharing is most often implemented in collaboration transparent systems, but can also be developed through collaboration-aware, special purpose applications. An important issue in application sharing is shared control. The primary design decision in sharing applications is to determine whether they should be centralized or replicated:

Centralized Architecture

In a centralized architecture, a single copy of the shared application runs at one site. All participants input to the application is then distributed to all sites. The advantage of the centralized approach is easy maintenance because there is only one copy of the application that updates the shared object. The disadvantage is high network traffic because the output of the application needs to be distributed every time.

Replicated Architecture

In a replicated architecture, a copy of the shared application runs locally at each site. Input events to each application are distribution to all sites and each copy of the shared applications is executed locally at each site.

The advantages of this architecture are low network traffic, because only input events are distributed among the sites, and low response times, since all participants get their output from local copies of the application. The disadvantages are the requirement of the same execution environment for the application at each site, and the difficulty in maintaining consistency.

Conferencing

Conferencing supports collaborative computing and is also called synchronous tele-collaboration. Conferencing is a management service that controls the communication among multiple users via multiple media, such as video and audio, to achieve simultaneous face-to-face communication.

Purposes in a Tele-Conferencing System

Video is used in technical discussions to display view-graph and to indicate how many users are still physically present at a conference. For visual support, workstations, PCs or video walls can be used. For conferences with more than three or four participants, the screen resource on a PC or workstation run out quickly, particularly if other applications, such as shared editors or drawing spaces, are used. Hence, mechanisms which quickly resize individual images should be used.

Conferencing services control a conference (i.e., a collection of shared state information such as who is participating in the conference, conference name, start of the conference, policies associated with the conference, etc.) Conference control includes several functions. Conference states can be stored (located) either on a central machine (centralized control), where a central application acts as the repository for all information related to the conference, or in a distributed fashion.

II. Session Management

Session Management is an important part of the multimedia communication architecture. It is the core part which separates the control, needed during the transport, from the actual transport. Session management is extensively studied in the collaborative computing area; therefore we concentrate on architectural and management issues in this area.

Architecture

Session management architecture is built around an entity-session manager which separates the the control from the transport. By creating a reusable session manager, which is separated from the user-interface, conference-oriented tools avoid a duplication of their effort. The session control architecture consists of the following components.

Session Manager

Session manager includes local and remote functionalities. Local functionalities may include: Membership control management, such as participant authentication or presentation of coordinated user interfaces; Control management for shared workspace, such as floor control, Media control management, such as intercommunication among media agents or synchronization, Configuration management, such as an exchange of interrelated QoS; and Conference control management, such as an establishment, modification and a closing of a conference.

Media agents

Media agents are separate from the session manager and they are not responsible for decisions specific to each type of media. The modularity allows replacement of agents. Each agent performs its own control mechanism over the particular medium, such as mute, un-mute, change video quality, start sending, stop sending etc.

Shared workspace Agent

The shared workspace agent transmits shared objects (e.g., tele-pointer coordinate, graphical or textual object) among the shared application.

Session control

Each session is described through the session state. This state information is either private or shared among all session participants. Dependent on the functions, which an application required and a session control provides, several control mechanisms are embedded in session management.

Floor Control

In a shared workspace the floor control is used to provide access to the shared workspace. The floor control in shared application is often used to maintain data consistency.

Conference Control

In conferencing applications, conference control is used.

Media control

This control mainly includes functionality such as the synchronization of media streams.

Configuration control

Configuration control includes a control of media quality, QoS handling, resource availability and other system components to provide a session according to user's requirements.

Membership control

This may include services, for example invitation to a session, registration into a session, modification of the membership during the session etc.

III. Multimedia Networking System

A multimedia networking system allows for the data exchange of discrete and continuous media among computers. This communication requires proper services and protocols for data transmission Multimedia networking enables distribution of media to different workstation.

Layers, Protocols and Services

A service provides a set of operations to the requesting application. Logically related services are grouped into layers according to the OSI reference model. Therefore, each layer is a service provider to the layer above. The services describe the behavior of the layer and its service elements (Service Data Units = SDUs). A proper service specification contains no information concerning any aspects of the implementation.

A protocol consists of a set of rules which must be followed by peer layer instances during any communication between these two peers. It is comprised of the formal (syntax) and the meaning (semantics) of the exchanged data units (Protocol Data Units = PDUs). The peer instances of different computers cooperate together to provide a service.

Multimedia communication puts several requirements on services and protocols, which are independent from the layer in the network architecture. In general, this set of requirements depends to a large extent on the respective application.

Physical Layer

The physical layer defines the transmission method of individual bits over the physical medium, such as fiber optics. For example, the type of modulation and bit-synchronization are important issues.

With respect to the particular modulation, delays during the data transmission arise due to the propagation speed of the transmission medium and the electrical circuits used. They determine the maximal possible bandwidth of this communication channel. For audio/video data in general, the delays must be minimized and a relatively high bandwidth should be achieved.

Data Link Layer

The data link layer provides the transmission of information blocks known as data frames. Further this layer is responsible for access protocols to the physical medium, error recognition and correction, flow control and block synchronization.

Access protocols are very much dependent on the network. Networks can be divided into two categories; those using point-to-point connections and those using broadcast channel,s sometimes called multi-access channels or random access channels. In a broadcast network, the key issue is how to determine, in the case of competition, which gets access is how to determine, in the case of competition, which gets access to the channel. To solve this problem, the Medium Access control (MAC) sub-layer was introduced and MAC protocols, such as the Timed Token Rotation Protocol and Carrier Sense Multiple Access with Collision Detection (CSMA/CD), were developed.

Continued data streams require reservation and thoughput guarantees over a line. To avoid larger delays, the error control for multimedia transmission needs a different mechanism than retransmission because a later frame is a lost frame.

Network Layer

The network layer transports information blocks, called packets, from one station to another. The transport may involve several networks. Therefore, this layer provides services such as addressing, internetworking, error handling, network management with congestion control and sequencing of packets.

Again, continuous media require resource reservation and guarantees for transmission at this layer. A request for reservation for later resource guarantees is defined through Quality of Service (QoS) parameters, which correspond to the requirements for continuous data stream transmission. The reservation must be done along the path between the communicating stations.

The requirements on the network layer for multimedia transmission are a provision of high bandwidth, multicasting, resource reservation and QoS guarantees, new routing protocols with support for streaming capabilities and new higher capacity routers with support of integrated services.

Transport Layer

The transport layer provides a process-to-process connection. At this layer, the QoS, which is provided by the network layer, is enhanced, meaning that if the network service is poor, the transport layer has bridge the gap between what the transport users want and what the network layer provides. Large packets are segmented at this layer and reassembled into their original size at the receiver. Error handling is based on process-to-process communication.

Transport protocols, to support multimedia transmission, need to have new features and provide the following function, semi-reliability, multicasting, NAK (Non-Acknowledgement)- based error recovery mechanism and rate control. First, we present transport protocols, such as TCP and UDP, which are used in the Internet protocol stack for multimedia transmission, and secondly we analyze new emerging transport protocols, such as RTP, XTP and other protocols, which are suitable for multimedia.

IV. Internet Transport Protocols

The Internet protocol stack includes two types of transport protocols.

Transmission Control Protocol (TCP)

Early implementation is of video conferencing applications were implemented on top of the TCP protocol. TCP provides a reliable, serial communication path, or virtual circuit, between is assumed to reside is an internet host that is identified by an IP address. Each process has a number of logical, full-duplex ports through which it can set up and use as full-duplex TCP connections.

Multimedia applications do not always require ful-duplex connections for the transport of continuous media. An example is a TB broadcast over LAN, which requires a full-duplex control connection, but often a simplex continuous media connection is sufficient.

During the data transmission over the TCP connection, TCP must achieve reliable, sequenced delivery of a stream of bytes by means of an underlying, unreliable datagram service. To achieve this, TCP makes use of retransmission on timeouts and positive acknowledgements upon receipt of information. Because retransmission can cause both out-of-order arrival and duplication of data, sequence humbering is crucial. Flow control in TCP makes use of a window technique in which the receiving side of the connection reports to the sending side the sequence numbers it may transmit at any time and those it has received contiguously thus far.

For multimedia the positive acknowledgment causes substantial overhead as all packets are sent with a fixed rate. Negative acknolwedgemnet would be a better strategy. Further, TCP is not suitable for real-time video and audio transmission because its retransmission mechanism may cause a violation of deadlines which disrupt the continuity of the continuous media steams. TCP was designed as a transport protocol suitable for non-real-time reliable applications, such as file transfer, where it performs the best.

User Datagram Protocol (UDP)

UDP is a simple extension to the Internet network protocol IP that supports multiplexing of datagram exchanged between pairs of Internet hosts. If offers only multiplexing and check summing, nothing else. Higher-level protocols using UDP must provide their own retransmission, packetization, reassembly, flow control, congestion avoidance etc.

Many multimedia applications use this protocol because it provides to some degree the real-time transport property, although loss of PDUs may occur. For experimental purposes, UDP above IP can be used as a simple, unreliable connection for medium transport. In general, UDP is not suitable for continuous media streams because it does not provide the notion of connections, at least at the transport layer; therefore, different service guarantees cannot be provided.

Real-time Transport Protocol (RTP)

RTP is an end-to-end protocol providing network transport function suitable for applications transmitting real-time data, such as audio, video or simulation data over multicast or unicast network services. It is specified and still augumented by the Audio/Video Transport Working Group. RTP is primarily designed to satisfy the needs of multi-party multimedia conferences, but it is not limited to that particular application.

RTP has a companion protocol RTCP (RTP-Control Protocol) to convey information about the participants of a conference. RTP provides functions, such as determination of media encoding, synchronization, framing, error detection, encryption, timing and source identification. RTCP is used for the monitoring of QoS and for conveying information about the participants in an ongoing session. The first aspect of RTCP, the monitoring is done by an application called a QoS monitor which receives the RTCP messages.

RTP does not address resource reservation and does not guarantee QoS for real-time services. This means that it does not provide mechanism to ensure timely delivery of data or guaranteed delivery, but relies on lower-layer services to do so. RTP makes use of the network protocol ST-II or UDP/IP for the delivery of data. It relies on the underlying protocols (s) to provide de-multiplexing. Profiles are used to specify certain parts of the header for particular sets of applications. This means that particular media information is stored in an audio/video profile, such as a set of formats (e.g., media encodings) and a default mapping of those formats.

Xpress Transport Protocol (XTP)

XTP was designed to be an efficient protocol, taking into account the low error ratios and higher speeds of current networks. It is still in the process of augmentation by the XTP form to provide a better platform for the incoming variety of applications. XTP integrates transport and network protocol functionalities to have more control over the environment in which it operates.

XTP is intended to be useful in a wide variety of environments, from real-time control systems to remote procedure calls in distributed operating systems and distributed databases to bulk data transfer. It defines for this purpose six service types: connection, transaction, unacknowledged data gram, acknowledged datagram, isochronous stream and bulk data. In XTP, the end-user is represented by a context becoming active within an XTP implementation.

V. Conclusions

Establishing a conference, where the conference participants agree upon a common state, such as identity of a chairman (moderator), access rights (floor control) and audio encoding. Conference systems may perform registration, admission, and negotiation services during the conference establishment phase, but they must be flexible and allow participants to join and leave individual media sessions or the whole conference. The flexibility depends on the control model, Closing a conference, and Adding new users and removing users who leave the conference.

VI.REFERENCES

[1]. Azim, M. A. Urban, D., Sayeeda, R.; Trends in medical education: Challenges and direction for need-basedreforms of medical training in South-East Asia, Indian Journal of Medical Sciences, 58(9): pp369-380 (2004).

[2]. Andleigh, P. K. & Thakur, K.: Multimedia systems design. PHI: ew Delhi (2003).

[3]. Bahera, A. P.: EDUSAT and its utilization. Retrieved June 21, (2010).

[4]. Bates, A. W.: Technology, e-learning and distance education, (2nd Edition) Routledge (2005).

[5]. Drury, P.: e-Health; a model for developing countries. E-Health International, 2(2):pp19-26 (2005).

[6]. Escobar, F.,Fraces, I., Bishop, I., & Zerger, A.: Evaluating a multimedia based tool for self-learning geographic information systems,June 20 (2010).

[7]. Gunjal, Bhojaraju & Urs, Shalini R.: Knowledge organisation systems, Part 1- comparative study of forty digital libraries, Proceedings of International Conference on Digital Libraries. New Delhi: TERI (2010).

[8]. Norris, P.: In Digital Divide. Cambridge, Cambridge University Press (2001).

[9]. Rajshekharan, K. & Nafala, K. M.: Digital archiving of audio content: Using WINISIS and Greenstone software. New Delhi: UNESCO (2009).

[10]. Vaughan, T.: Multimedia: Making it works (7th ed.). New Delhi: Mac-Graw Hill (2008).

[11]. Zhang, W.: Multimedia, technology, education and learning. November 10, 2007 (1995).

Cite this Article

Nabajyoti Sarma, Dr. J. P. Tripathi, "Study of Application Sharing Approach of Social Media Networking Sites", International Journal of Scientific Research in Science and Technology (IJSRST), Online ISSN : 2395-602X, Print ISSN : 2395-6011, Volume 10 Issue 1, pp. 713-718, January-February 2023.

Journal URL : https://ijsrst.com/IJSRST52310423