WebCon:
A personal conference system utilizing the World-Wide Web and the Multicast backbone

Kåre Synnes, Peter Parnes, Mattias Mattson, Dick Schefström
Centre for Distance-spanning Technology / Departement of Computer Science
Luleå University, Sweden

{Kare.Synnes, Peter.Parnes,Mattias.Mattsson,Dick.Schefstrom}@cdt.luth.se

Abstract

This document is a position paper for a workshop organized by the World Wide Web Consortium on "Real Time Multimedia and the Web" (RTMW '96).

The paper presents a personal conference system, which integrates real-time media transport through the Real-time Transfer Protocol into a World-Wide Web based environment. The Session Invitation Protocol is used for user location resolution, user invitation and session initialization. The session control and synchronization is done through usage of the WebDesk Control Bus, which in turn uses the Scalable Reliable Real-time Transport Protocol for distribution of control messages.

The WebCon is deployed as a study for future ways of personal conferencing within the EC founded MATES project.

Introduction

Communication over the Internet is today growing rapidly, deployed and fueled by the high costs and complexity connected to the traditional ways of communication using telephone equipment. Several tools for Internet telephony has been announced during the last months and video conferences over the Multicast Backbone (MBone) [1] are becoming more and more popular.

The World-Wide Web (WWW) [2] in conjuction with real-time media distribution through the Real-time Transfer Protocol (RTP) [3], offers an unique possibility for bringing low-cost personal conferencing to anyone able to use a Java [4] enabled WWW browser.

The WebCon intends to fill the need for such a system, while bridging the gap between the WWW and the MBone.

The WebCon

The traditional way of tele-communication is that of a tightly controlled session, as opposed to the loosely controlled sessions now used in the MBone world. For personal conferencing there is a need for tightly controlled sessions, since the matters discussed most often can be considered private.

The WebCon is a personal conference system for easy-to-use video-phone services over the Internet, using a tightly controlled scheme for participation. The implementation of the WebCon can be divided into three subsystems:

- the WebCon client (client),
- the WebCon server (server) and
- the WebCon agent (agent).

The WebCon server

The WebCon server is a platform dependent audio/video server that interfaces to the hardware devices. It is not a central server but local to each host and user. The server is needed for sharing of resources, as on some platforms only one may access the hardware at a time.

The server is implemented in C for efficiency and exchanges raw data with the client over a set of sockets: Audio in (automatically mixes all inputs), Audio out and Video out.

The WebCon client

The WebCon client contains the graphical user interface (GUI) and the real-time media transport and control protocols. The client is implemented as an Java Applet for platform independency and executes within a WWW browser giving the user a dynamic interface to the conference controls.

The client GUI contains one window for video output, a list of participants and controls to achieve invitation, video and audio control (see Figure 1).

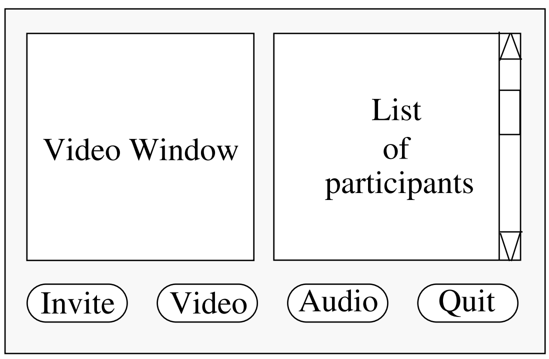

Figure 1, the WebCon client.

The video stream showed is either selected by the user or switched automaticly accordingly to whom is currently speaking. Redundant audio encodings will be supported, in order to recover from packet-loss

The client communicates with the agent over the WebDesk Control Bus (CB) [5] and with the server using the described set of sockets. The CB uses the Scalable Reliable Real-time Transport Protocol (SRRTP) [6] for messaging (SRRTP is the RTP extended with the ideas from the SRM framework [7]).

The WebCon agent

The WebCon agent is an Java application that handles user location resolution, session setup etc through the Session Invitation Protocol (SIP) [8]. SIP is a simple protocol designed to enable the invitation of users to participate in multimedia sessions, such as in a personal conference system like the WebCon. SIP also enables user mobility by relaying and redirecting invitations to the user's current location.

SIP is used together with an universal communication identifier (UCI) for session setup. An UCI is the equivalent of a personal telephone number, for the WebCon the location-independent email-adress is chosen as UCI.

The UCI is resolved into a more precise identifier, namely a username plus machine host name (unicorn@druid.cdt.luth.se) where the user is located, by mechanisms in the SIP protocol. This is used to setup the conference between two or more participants. The communication between the three parts is showed below (see Figure 2).

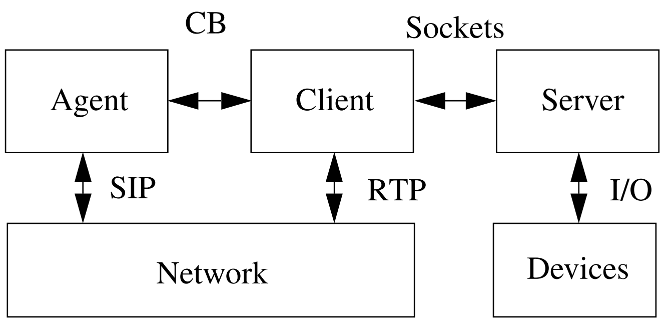

Figure 2, the WebCon system.

There is however no way of protecting oneself against unwanted listeners, since the usage of multicasting allows anyone to 'tune-in'. Encryption and/or authentication and call screening should therefor be included in the WebCon as described in [8].

A WebCon session

A calling party selects a WWW page with a client. This automaticly starts the server and agent (if the browser cannot spawn them itself, then these must be initialized manually).

The invitation is done by the calling party through selecting a entry from a personal list with possible participants. The agent handles the session setup and the audio, video and control flows are started if the called party acknowledge the invitation. The called party must have an agent running, otherwise the invitation will fail.

The participants in a session may invite additional participants. This is simply done the same way as the initial invitation. The session is terminated when one participant remain in the session.

Problems

The access to the filesystem and sockets from within a browser may be limited, because of security restrictions. E.g., Java has a security manager that may deny access to the resources on the local host. One solution is that the browser only starts the WebCon, which instead runs as an application. Another solution is to supply an own security manager for Java.

Implementation

The WebCon is currently beeing implemented within the MATES framework, as a part of the MATES WebDesk [9] framework for distributed collaboration with audio, video and shared applications.

Acknowledgments

This work was done within the Esprit project 20598 MATES, which is supported by the Information technology part of the 4:th Framework Programme of the European Union.

References

[1] The MBONE information web, <URL:http://www.mbone.com>

[2] Berners-Lee, Cailliau, Luotonen, Nielsen & Secret, *The World-Wide Web*, Communications of the ACM, August 1994

[3] H. Schulzrinne, S. Casner, R. Frederick, V. Jacobson, *RTP: A Transport Protocol for Real-Time Applications*, Internet RFC 1889, January 1996

[4] JavaSoft Inc., Java, technical Report, 1996, <URL:http://www.javasoft.com/>

[5] P. Parnes, *Control Bus Specification*, 1996, <URL:http://www.cdt.luth.se/~peppar/docs/>

[6] P. Parnes, *Scalable Realiable Real-time Transport Protocol - SSRTP*, work in progress, <URL:http://www.cdt.luth.se/~peppar/docs/>

[7] Floyd, S., Jacobson, V., McCanne, S., Liu, C., Zhang, L., *A Reliable Multicast Framework for Light-weight Sessions and Application Level Framing*, ACM SIGCOMM'95

[8] M. Handley, E. Schooler, *SIP: Session Invitation Protocol*, Internet Draft: draft-ietf-mmusic-sip-01

[9] P. Parnes, D. Schefström, K. Synnes, *WebDesk: the Collaboration support in MATES*, First European Workshop on Global Engineering Network, 1996, <URL:http://www.cdt.luth.se/~peppar/docs/>