Secured Video Conferencing Desktop Client for Telemedicine

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Abstract- **As a result of growing interest in telemedicine during the last decade, we have seen the development and deployment of several video conferencing telemedicine applications during recent years. Deployment of these applications was on ISDN lines or dedicated circuits since the bandwidth, security, and privacy have been the biggest concern in telemedicine. Session Initiation Protocol (SIP) is a signaling standard that attracts telemedicine application developers since SIP was designed for multimedia communications over IP-based networks and has inherent native mode security mechanisms built in.**

We have developed a SIP based video conferencing application that runs over the Internet and can provide authentication using MD5 Digest hashing mechanism. This client is designed to provide voice and video services using SIP, currently being standardized by IETF. The client is also integrated with directory service designed on commObject architecture. The video conferencing client performance and interoperability has been successfully tested. We are now working on deploying it in a telemedicine environment. In a telemedicine setting we will gather data from users in order to evaluate user satisfaction for both physicians and patients. There are additional features we have identified that can enhance the patient-physician communication and they will be obtained easily from our client in the future.

I. INTRODUCTION

As a result of growing interest in telemedicine during the last decade [1], we have seen the development and deployment of several telemedicine applications during recent years [2]. Telemedicine has various potential uses like educational, administrative and clinical [3]. However, one particular area has been more popular than the others in US: high quality, interactive video-consultation between remote rural patients and tertiary-care specialists [3]. The applications such as remote patient monitoring, consultation, and diagnosis have been developed [1] that runs over ISDN lines which is often very costly. However, the patients that demand home health services could not benefit from existing technologies until the widespread adoption of Internet. Internet enables the distribution of health services with low cost. Many studies have been conducted to measure patient satisfaction with telemedicine in homecare systems and the results indicate that the majority is satisfied by the low-cost low-bandwidth solutions [4].

The biggest concern in telemedicine is still the security and privacy of the patients [5]. Necessary security mechanisms, which provide authentication and authorization, are necessary to conduct telemedicine over the Internet. Privacy and confidentiality using encryption is also desirable. A recent Voice over IP signaling standard approved by IETF called Session Initiation Protocol (SIP) attracts telemedicine application developers since SIP can handle voice, video as well as multimedia communications over IP-based networks and has inherent native mode security mechanisms built in [6]. Existing telemedicine applications are based on the ITU standard H.323. However, this standard does not lend itself to integration with web, messaging and does not have a native security mechanism. Additional effort and integration with other security mechanisms is necessary to provide authentication and authorization.

A SIP based video conferencing application, CGUsipClientv1.1, has been developed that can provide authentication using MD5 Digest hashing mechanism. This application is designed to provide voice and video services using SIP, currently being standardized by IETF. In section II, we will first present the architecture of the SIP based video conferencing application developed. Section III explains the features of the application that are useful for telemedicine and provide the results of interoperability tests with other video conferencing products. The paper concludes with the challenges we face in telemedicine and how this application can be used as a telemedicine solution.

II. SIP BASED VIDEO CONFERENCING CLIENT

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A. Architecture

Session Initiation Protocol (SIP) is the Internet Engineering Task Force (IETF) standard for IP Telephony. It is an application layer control protocol that can create, modify, and terminate multimedia sessions [6]. Different types of entities are defined in SIP: user agents, proxy servers, redirect servers, and registrar servers. Fig. 1 shows a simple SIP call flow including these entities.

Although there are open source SIP stacks such as Vovida and NIST, Dynamicsoft commercial stack was used to develop CGUsipClientv1.1. Dynamicsoft provides a comprehensive SIP stack including all the authentication mechanisms included in the latest RFC [6] which are not available in open source user agent SIP stacks. The client is designed to work with any SIP proxy, open source or patented, and interact with other SIP user agents.

Java Media Framework (JMF) 2.1.1 Sun libraries were used to develop a SIP based video conferencing application. JMF 1.0 API (the Java Media Player API) enabled programmers to develop Java programs that presented time-based media. JMF 2.0 API extended the framework to provide support for capturing and storing media data, controlling the type of processing that is performed during playback, and performing custom processing on media data streams [8].

CGUsipClientv1.1 architecture is presented in Fig. 2. There are two main Java packages – *cgusip.client* and *cgusip.utils* – that structure CGUsipClientv1.1 [9]. The *utils* package handles the existing instances of sip connections and calls. The client package has three main components: *gui*, *sip*, and *media*. The *gui* package handles all aspects of the client user interaction. The *sip* package handles the necessary interaction between the CGUsipClientv1.1 and the Dynamicsoft sip stack for creating and terminating sessions and initiating and receiving calls. The *media* package is in charge of making media connections using JMF and Dynamicsoft SDP libraries.

B. Supported Audio/Video Codecs

It is important to support established standards in video conferencing for video and audio. This prevents the interoperability problems between various clients. Therefore, CGUsipClientv1.1 supports three widely used standard video formats: h.263, h.261, and jpeg. H.263 is the default for video sessions. Supported audio formats are:

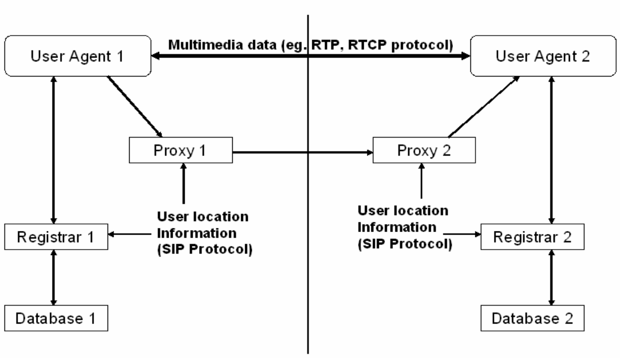

Figure 1. A typical SIP configuration [modified from 7]

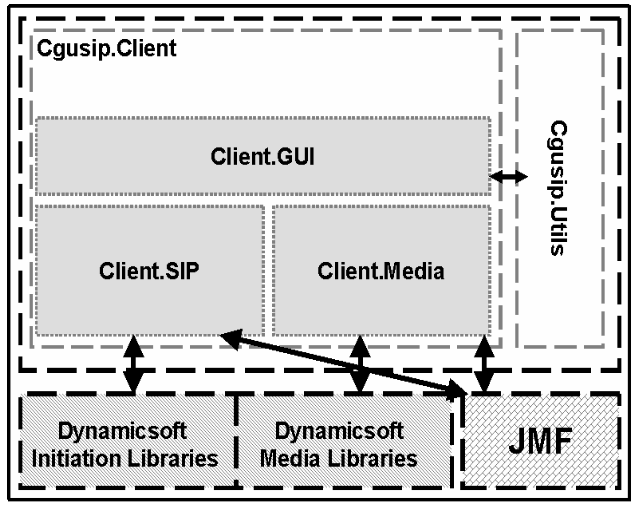

Figure 2. CGUsipCleintv1.1 architecture [9]

- g.723 and GSM: They are best suited for PC-to-telephone and PC-to-PC calls and used when the speed is 56K or less;
- DVI and µLAW: They are best suited for PC-to-PC calls and should not be used if the speed is less than 56K.

The default audio codec is g.723. The codecs used in this application enable us to provide a low-cost application to video conferencing users.

C. Interoperability Test Results

CGUSipClientv1.1 had been successfully tested for point-to-point voice and video communication [9]. There was a need to test the client with other clients and proxies after initial period of distribution. CGUsipClientv1.1 was able to establish audio communication with MSN Messenger and Ubiquity's Helmsman User Agent. CGUsipClientv1.1 could not communicate with Siemens SIP user agent. CGUsipClientv1.1 and MSN Messenger could exchange voice but not video since MSN Messenger does not support the standard video codecs, but instead uses media7 proprietary codec.

III. SIP Based Video Conferencing Client Features

A. Low-Cost Video Conferencing

In telemedicine various complex video conferencing tools have been developed to provide high quality video conferencing for telediagnosis. However, we have not seen many applications that provide low-quality/low-cost video conferencing applications for home care and elderly care. As the Internet becomes more popular among all age groups, the need for such applications also increases. CGUsipClientv1.1 is a tool that promises low-cost video conferencing for home Internet users.

Either by registering to a SIP proxy/registrar or by typing a direct IP address, one can make multiple calls (up to 5 lines are available) to any user capable of receiving SIP calls. The tool consumes moderate amount of CPU power (40-50% during the call where both audio and video is in use). During a conference call where both audio and video are in use, it consumes 50-80 Kbits per second. This is achieved through the use of standard audio-video codecs that are suitable for low-bandwidth connections. Chae et al. [4] conducted a study on evaluating the low-bandwidth telemedicine systems that are used for elderly home care. They concluded that a majority of elderly patients were satisfied with low-bandwidth telemedicine.

Using this tool a patient, who has Internet access at home, can simply make a call to his/her physician or nurse for cases where there is no need for high quality video connection between these two users. Fig. 3 shows the screen snapshots of two users in call. An example use of this tool may be home telemedicine for diabetes. More than 90 million Americans [10] are living with chronic illnesses, such as diabetes hypertension, etc. Mun and Turner [10] stated that the patient care protocol for chronically ill patients is well established. Therefore, these protocols can be managed remotely if the patients have necessary equipments at home. CGUsipClientv1.1 enables the physicians and nurses to monitor the patients remotely through a regular Internet connection.

B. Security

Security and privacy are among the most critical problems of telemedicine over the Internet [5]. For achieving large number of video conferencing users over the Internet it is mandatory to provide secure authentication and authorization mechanisms with the applications. Two main security mechanisms used with SIP are authentication and data encryption [11]. Data authentication is used to ensure that the doctors sending the messages are who they claim to be. It is also used to make sure that message information was not modified during the transit [11]. Data encryption, which protects the confidentiality of the communication, is used to ensure that only the intended person can decrypt and read a message [11]. In order to provide authentication service both the servers and the clients involved in the call process have to support these security mechanisms.

In our testbed, dynamicsoft proxy and registrar provides authentication for users. Authentication type, which could be basic or digest [11], is assigned to each user during the registration process. For each registration request, registrar challenges the users for authentication based on the authentication type. The user agent (CGUsipClientv1.1) is responsible for requesting the necessary information from the user, such as username and password, and sending it to the registrar for authentication. Proxy also challenges each user for all requests except ACK and BYE message. This ensures that the user who is making a call is the person who they claim to be. When a user agent is challenged, it would have to resubmit its request with security credentials. The request and credentials will be verified using digest mechanism by proxy against those in the location server [12]. Further, location server would verify against the database in which it stores the details of users.

Figure 3. CGUsipCleintv1.1 snapshots

In a telemedicine scenario, if a patient wants to reach a physician, they first need to register with their username and password to a registrar/proxy server. This process is mandatory even the user wants to dial a direct IP using CGUsipClientv1.1. During the registration user is challenged by the server and necessary credentials are exchanged based on the Digest mechanism. The patient can now make a call to any registered users or to a direct IP address. The information about the user's identity will be transferred to the physician via Caller ID function.

Fig. 4 shows a snapshot of the receiving end. The CallerID window pops up when the physician receives a call. Caller information that is retrieved from the database is provided to the receiving end only if the patient is authenticated. We call this a knowledge-management caller ID. Doctors work with numerous patients. But when a patient calls a doctor, if the caller ID can identify who the patient is, provide some more medical information, it helps the doctor to provide customized service. In fact an option in this pop up could be to retrieve more information from patient database. User can either take the call by clicking the Answer button or they can drop the call by clicking the Busy button. Other options that a user can explore in the CallerID window are: (1) user can add the caller to his/her Private Address Book (PAB) by clicking the AddtoPAB button or (2) user can redirect the call by clicking Redirect (see next section for details).

Figure 4. CGUsipCleintv1.1 CallerID feature

Data encryption mechanisms have not been implemented in CGUsipClientv1.1. Next version of CGUsipClient will implement an enterprise-wide authentication mechanism with single-sign-on. Digital certificates are being explored for this purpose.

C. Redirect Service

A physician may want to redirect a call to another colleague for expert advice. If the colleague also has a sip user agent, the call can be redirected without answering it. The call can also be redirected to other devices such as cell phone, a local office phone, or voice mail server. A SIP/PSTN gateway is required in order to redirect a call to a local or mobile phone. The redirecting physicians can also choose to redirect a call to a particular web site which may contain medical information. In that case, the callee would see a pop up window open on his/her PC and can then choose to go to that site. Fig. 5 illustrates the redirect process from the physician's perspective.

When the patient receives the redirect, he/she can select from the options provided for redirect and the client will handle the necessary process to fulfill user's request. Fig. 6 shows the redirect screen that a caller receives after the physician redirected him/her to a different destination.

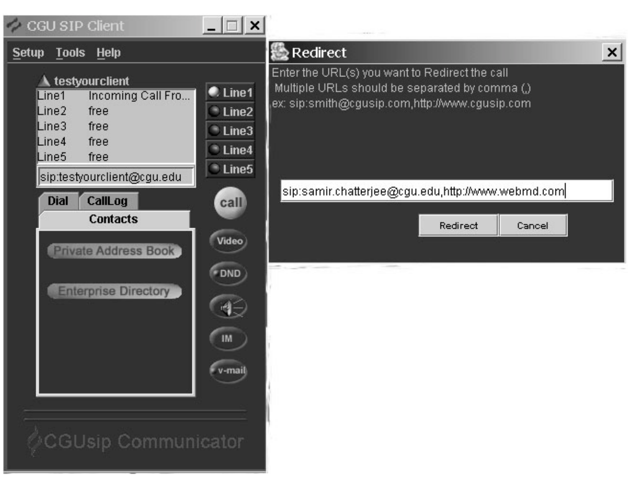

Figure 5. CGUsipCleintv1.1 Redirect feature – Call received

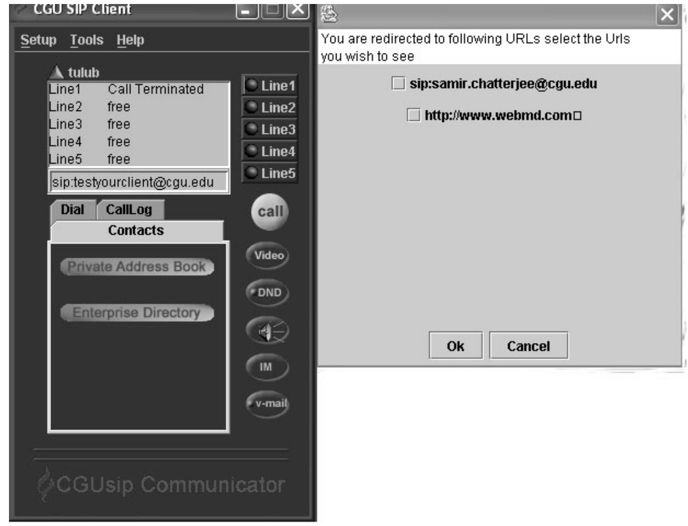

Figure 6. CGUsipCleintv1.1 Redirect feature – Redirect Received

D. Directory Services

In telemedicine, it is important for physicians to look up other doctors, nurses or specialists. This is typically done in a white page lookup which is often implemented via a directory service. It should be simple to use and easy to search. Lightweight Directory Access Protocol (LDAP) was first proposed in the early 90's [13]. LDAP structures the information using a systematic scheme that is designed in a hierarchical manner. LDAP has been designed for optimized directory lookups.

CGUsipClientv1.1 is integrated with directory service designed on commObject, an architecture that is recommended as a draft [14] by ViDeNet group of Internet2 community. CommObject architecture is designed for storing protocol specific attributes that integrates with white page information in enterprise directory. CommObject architecture suggests objects that need to be inherited by enterprises, which plan to implement directory structure for voice or video communications. The architecture can be applied to other video-conferencing applications such as VRVS on UNIX, H.323 or SIP.

CGUsipClientv1.1 integrates the architecture to involve SIP related attributes. These attributes can be defined to include details of patients, physicians, nurses, administrative staff, or groups. SIP attributes, such as SIPURI, SIPRegistrar, SIPProxy, are defined in commObject architecture until now. The architecture for SIP is still under discussion.

IV. CHALLENGES

Brown et al. [15] reported that telemedicine affects three major challenges to healthcare: access, cost, and quality. Increasing access, reducing cost, and improving quality of healthcare became possible through telemedicine technologies.

We believe that our video-conferencing software will vastly improve access to telemedicine services since it does not require any expensive set up and uses the Internet for transporting voice and video information. Our client software works well with standard off-the-shelf cameras and microphones.

Today's Internet does not have Quality-of-Service (QoS), but rather provides a best effort service. In such a service, there are no bandwidth guarantees, packets can get lost and delayed. However, through extensive testing over regular IP networks, we have achieved reasonable to good performance without QoS. The video frame size is limited to Quarter Common Intermediate Format (QCIF) (176 pixels by 144 lines at 30 frames per second) which is small. But in our software, we can drag and enlarge the screen size but we begin to lose fidelity. We are exploring wavelet based codecs which can give us better performance in such cases than the Discrete Cosine Transformation (DCT) codecs that we currently use.

Security is always a critical challenge for healthcare information. We do not want to communicate with the criminals who may pose as doctors and neither do we want our medical data to be snooped by others. Our client provides an effective authentication mechanism. We are currently building role-based authorization within the system so that only authorized users can access medical records. At a future point we want to include encryption using digital certificates. However, there are still issues of using a Public Key Infrastructure (PKI). Most applications use server size certificates and client certificates are still not popular. Then there are issues with having a Certification Authority (CA) and taking care of revocation lists. But with recent approval of the Health Insurance Portability and Accountability Act of 1996 (HIPAA) regulation, it is likely that hospitals and medical environments would provision for such infrastructure.

Video-conferencing between multiple end-points enables collaboration environment in telemedicine. To make multiple end-points communicate with each other, a Multiple Conferencing Unit (MCU) is required. This unit does the work of mixing audio and video signals and transferring the resultant signals to the end-point. As of now, SIP-based MCUs are still under development in the industry such as Session™ Conference Server Software by Wave3 and viaIP MCU by Radvision. Currently, CGUsipClientv1.1 provides a point-to-point conferencing interface with no additional components. Multi-point conferencing through SIP-based clients would be possible as SIP-based MCUs become widely available.

V. Conclusion

In this paper, a SIP-based video conferencing client was introduced. The client architecture, features, and interoperability issues were discussed. We are now working on deploying it in a telemedicine environment. In a telemedicine setting we will gather data from users in order to evaluate user satisfaction for both physicians and patients. The additional features we have explained in this paper can enhance the patient-physician communication and more useful features will be obtained easily from our client in the future.

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