Project 5: Voice over IP

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1 Introduction

The Internet is becoming the ideal platform for all forms of modern communication including the newly developed IP telephony systems based on the SIP and H323 standards and non-Internet based applications such as wireless applications. H.323 was proposed by the ITU in 1998 as a complete protocol suite while the Session Initiation Protocol (SIP) was designed by the IETF and standardized in RFC3261 as an application-layer protocol which describes how to set up Internet telephone calls, video conferences, and other multimedia connections. SIP is a single module designed to internetwork with existing Internet applications by defining telephone numbers as URLs, so that telephone communication can be initiated through Web pages by allowing a mouse click to initiate a telephone call. SIP can run over UDP or TCP to handle the setup, management, and termination of sessions; letting the task of data transport for other protocols such as RTP/RTCP.

2 General Description

SIP is a protocol that (1) supports a variety of services including locating the called and determining the callee's capabilities, as well as handling the mechanics of call setup and termination and (2) provides the capability to create new services such as call center or audio conference emulation.

SIP-based IP telephony has several advantages, many of them resulting from the ability to run the server and the clients on separate machines on the Internet. These include:

- Integration of transport: different businesses run on top of different technologies
- Cost savings: long distance call at the price of local calls
- Open service market: anyone with IP connectivity can become provider
- Programmability: split of transport and application services allowing each person who has connectivity to become provider
- Integration of applications: split of applications allowing for example IP to be operated by an operator, SIP signaling to be operated by another operator and PSTN call termination by a third operator.

The NIST SIP is a voice over IP kit that (1) enables the measurement of voice quality for existing VoIP end systems and networks (2) facilitates the analysis of new VoIP coding and representation technologies and network transport services and (3) expedites the research and development of protocol sand platforms for programmable telephony services.

3 Project description

This project serves as an introduction to Voice over IP programming. You will be expected to understand how a Voice over IP System works and program a voice over IP toolkit to add new services to a voice over IP system. This project includes two parts: (1) the investigation of three VoIP examples and (2) the extension of the SIP kit to add services to a VOIP system.

3.1 Investigation: understanding how a VoIP system works

NIST SIP provides three examples illustrating how SIP can be used to implement (1) a soft phone (JsPhone example), (2) a proxy (Proxy example) and (3) an instant messaging client (Instant messaging example). Students will use the NIST SIP kit installed on the network to investigate the three applications examples provided by the kit. For each example, describe (1) what the package does, (2) the type of SIP messages used, (3) the main SIP components involved in the example, (4) how to start the example, and (5) how it works. This part of the project project will be assessed based on (1) a presentation of experiments done using the SIP kit and (2) a report on the investigation of VoIP containing the elements defined above.

3.2 Extension: adding services to a VoIP system

The basic NIST SIP kit needs extensions to be used in a modern communication environment requiring functionalities such as secure voice and data transmission, billing, video-conferencing, call and contact centers, VPNs and hybrid access to Internet-based and non-Internet based applications. We consider an extension to the NIST SIP software package for adding authentication and billing to a VoIP system.

1. User authentication is an important process in IP telephony which protects the system from unauthorized users by checking the calling party's IP address against an authentication database. This authentication database may also be useful in billing by defining access rights to different services provided by the IP telephony system so that each user may be authenticated and its service access rights evaluated.

2. Billing is also an important component of a telephony system which will be implemented in the NIST SIP kit by a third-party billing system using time-stamped information about every processed call and service access rights provided by the authentication database to assess the service access rights provided to each user.

This part of the project project will be assessed based on (1) the architecture of the extended VoIP system and (2) the correctness of the program.

References

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[5] Java Network Programming Tutorial, `java.sun.com`