NORTHERN

IMPLEMENTATION OF AN INTERNET PROTOCOL TELEPHONY SYSTEM

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Abstract

Internet Protocol Telephony is a group of technologies that allows to make telephone calls over Internet Protocol networks instead of the traditional public switched telephone networks (PSTNs). Internet Protocol Telephony provides several benefits. It allows multimedia communication, and can reduce or eliminate the fees charged by public switched telephone networks (PSTN) providers.

Motivation

Our interest in Internet Protocol Telephony is highly motivated by its cost savings option since it avoids the tolls charged by traditional telephony services, and its fairly ease of deployment. We are interested in learning more about this technology in order to make a positive impact in the telecommunication sector.

Introduction

A protocol is a set of rules that describes data delivery in telecommunication networks.
Internet Protocol Telephony employs a variety of protocols, including including Real Time Protocol (RTP) for the transport of voice and multimedia data, and Session Initiation Protocol (SIP) to establish, maintain and terminate sessions.
A basic Internet Protocol Telephony system is composed of a SIP server, a gateway and end-devices. The SIP server provides centralized call set-up and call management functions. The gateway handles routing to different domains and signal translations between traditional telephone equipment and IP network devices. End-devices are user agents and can be a mix of IP capable phones, traditional telephones, and computer-based softphones.

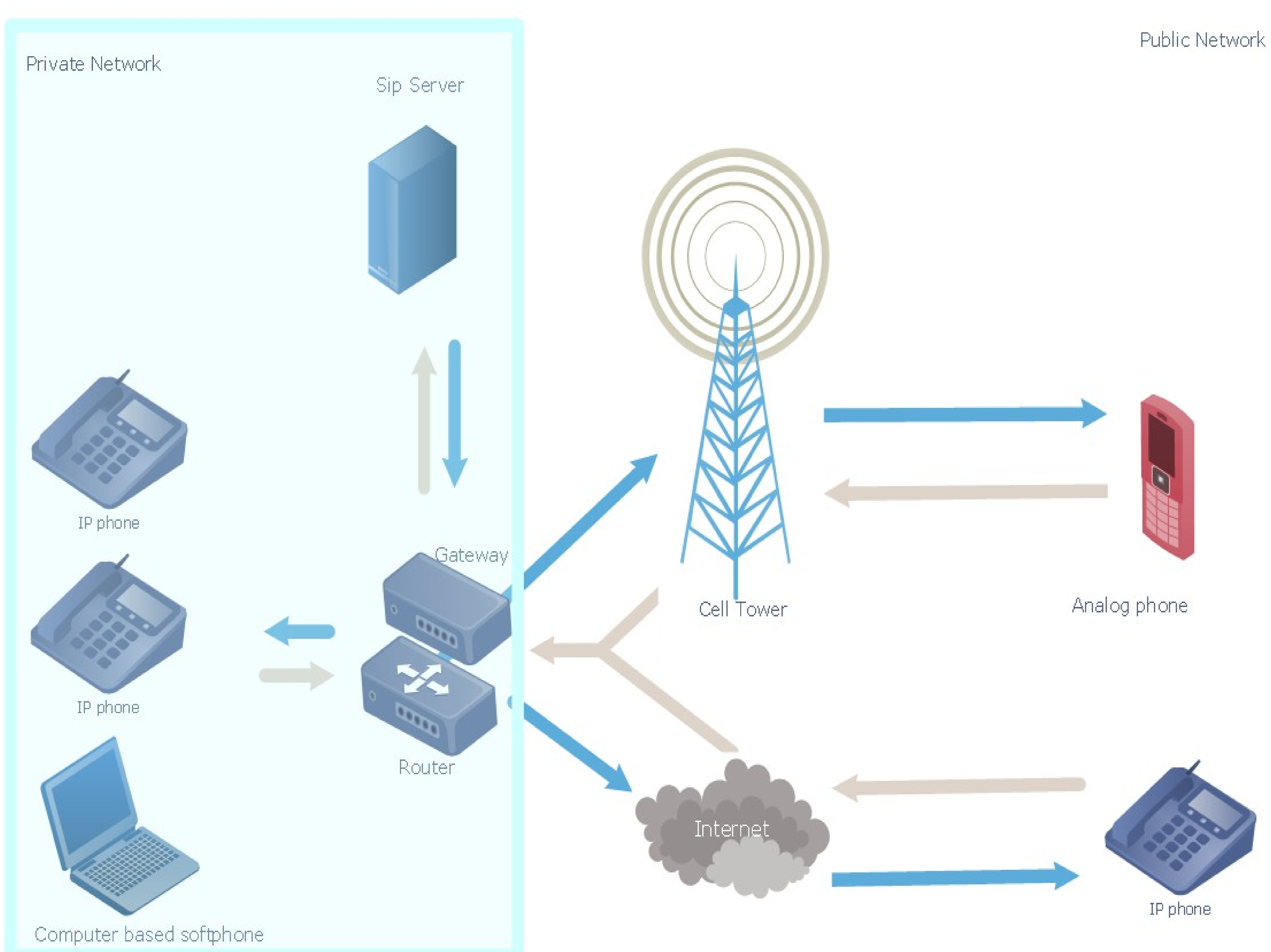
Goals for the Project

- Gain a deeper understanding of the SIP protocol
- Implement support for users registration
- Implement support for users authentication
- Implement support for network address translation

Methods and Materials

- One computer with Linux operating system
- One switch
- One hub
- One router
- One PSTN gateway
- Three IP-Phones
- Kamailio SIP server
- MYSQL database
- Jitsi Sip communicator
- Wireshark protocol analyzer

Network Diagram



Process

User agents are assigned with user-names and passwords for the SIP server domain, and pre-recorded in the MYSQL database. For user agents to start communicating with each others, they must first register with the SIP server and be authenticated.
To register with the server, a user agent sends a REGISTER request to the server which obtains the IP address of the user agent, and associates it with the user-name and password of the user agent in the MYSQL database.
Authentication is a process in which credentials provided by user agents are compared to those of authorized users in the database. Authentication happens during REGISTER and INVITE requests in order to secure our SIP server. This prevents anonymous users to have the ability to register with our SIP server, and prevents unauthenticated users to make telephone calls.
Network address translation (NAT) operates on a router and enables private IP networks that use unregistered IP addresses to connect to the Internet. It translates private addresses in the internal network into public addresses before packets are forwarded to the public network.
The gateway handles routing to different domains and signal translations between traditional telephone equipment and IP network devices.

Conclusions

Working on this project mostly required Sip protocol knowledge, networking knowledge, database knowledge, and Linux operating system knowledge.
We were successful in implementing an Internet Protocol Telephony System with support for users registration, users authentication, and network address translation.
With our current configuration, users from our private network can call each other, call and receive calls from users on the internet.

Future Work

- Implement routing to public switched telephone networks (PSTNs)
- Implement call forwarding.
- Create a Java application to keep track of SIP resquests and responses processed by the SIP server.

References

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