OpenVoIP: An Open Peer-to-Peer VoIP and IM System

Salman Abdul Baset, Gaurav Gupta, and Henning Schulzrinne
Dept. of Computer Science, Columbia University
New York, NY, USA
{salman,gg2265,hgs}@cs.columbia.edu

ABSTRACT

This demo presents OpenVoIP, a 500 node open peer-to-peer VoIP and IM system running on Planet Lab. The three key aspects of OpenVoIP's design are its ability to use any DHT or unstructured peer-to-peer protocol for directory service, the use of intermediate peers with unrestricted connectivity to relay signaling and media traffic between peers behind NAT and firewalls, and a diagnostic system integrated with Google maps for graphical monitoring. The demo will show these aspects of OpenVoIP and provide insights on issues and related problems in building such a system.

Categories and Subject Descriptors: C.2 [Computer Communication]: Network Protocols, Network Architecture and Design

General Terms: Protocols, Architecture, Measurement.

Keywords: Peer-to-Peer, VoIP.

1. INTRODUCTION

The research in peer-to-peer systems has focused on the design of structured or unstructured protocols, file sharing, and streaming. Skype is the first peer-to-peer application in the area of VoIP that provides a distributed directory service for locating usernames, establishes VoIP and media sessions between nodes, and uses machines with unrestricted connectivity to relay signaling and media traffic for nodes behind NAT and firewall. Recent study has shown that Skype call success rate depends on the characteristics of a network connection such as NAT, and it does a poor job in finding a nearby relay between caller and callee [4]. Also, our conversations with Skype users suggest that at times they became annoyed with Skype using their machine resources for relaying media sessions and decided to shut down the Skype application. This results in drop calls.

To systematically explore the issues in a peer-to-peer VoIP and IM system, we have designed and built OpenVoIP [1], a 500 node peer-to-peer VoIP and IM system, and deployed it on Planet Lab. The goals of this demo are (1) to show the live system (2) to allow nodes to join and leave OpenVoIP, thereby generating realistic churn (3) to use any DHT or unstructured protocol to locate user names (4) to establish media and IM sessions between nodes in the presence of NATs and firewalls by using another machine as a network relay (5) to demonstrate a diagnostic and monitoring system integrated with Google maps that keeps track of node state and shows lookup request progress.

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2. SYSTEM DESIGN AND ARCHITECTURE

OpenVoIP's design goal is to potentially use any DHT or unstructured protocol for implementing a directory service, allow nodes with unrestricted connectivity to relay signaling (SIP) and media (RTP) traffic between nodes behind NAT and firewalls, and a diagnostic mechanism to quickly find faults in the overlay. To accomplish this, we have designed an application layer protocol which we refer to as Peer-to-Peer Protocol (P2PP) [3]. The design of P2PP exploits commonalities in the well known structured and unstructured protocols such as Chord, CAN, Kademlia, Bamboo and Gia thereby defining a protocol that does not contain any of the above protocol specific details and has an extension mechanism to incorporate a protocol-specific feature. The protocol allows exposing the peer state such as routing table, disk usage, link capacity, bandwidth and CPU utilization. It defines mechanisms for NAT and firewall traversal, uses a secure transport, and has mechanisms for exchanging node capabilities and diagnostic information. It allows non-participant nodes to use overlay services through participant nodes. Our current implementation of P2PP supports three DHTs, namely, Chord, Kademlia and Bamboo.

We have integrated our protocol with OpenWengo [2], an open source SIP phone. This p2p phone connects to our 500 node OpenVoIP system running on Planet Lab. In the absence of NAT, the phone fully participates in our Planet Lab p2p network. If behind a NAT, the phone uses the services of our p2p network through a participant node. It uses the p2p layer to search for user names and uses SIP to establish media sessions between nodes. Overall, our system consists of a bootstrap server, a diagnostic server that keeps track of online nodes and updates their status on Google maps, and peers running our p2p executable or p2p phone. At present, we are using this system to investigate the issue of relay selection in p2p VoIP systems.

3. REFERENCES

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