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Design and validation of a methodology for distributed relay service for NAT traversal in a peer-to-peer VoIP network

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Introduction

The evolution of computer networks leaded to the deployment nearly omnipresent broadband IP-based networks. The cheapness and increasing performance of data transmission through the Internet favored the development of new services over this infrastructure. Those services are often in substitution and enrichment of traditional one, which are typically provided over dedicated circuits.

One example is *Television-over-IP* (or *IpTV*): the usage of the Internet as a transmission channel, instead of the common broadcast channels (air, cable, satellite) added interactivity to television. This means letting the user chose what watch and when (Video-on-Demand), but also let the user influence the program (e.g.: voting).

What is under the focus of this work, however, are the technologies that are leading to the gradual moving of the traditional circuit-switched phone line (*PSTN*[1]) over IP-based networks: *VoIP* (*Voice over IP*), also called *IP telephony*.

This transition is happening at different levels. Under the eyes of the public, software applications that emulate a traditional phone are very popular, and often distributed by a provider that is offering interconnection with the PSTN. Less evidently, some providers are pushing hardware devices that route analog voice signal coming from a traditional telephone, through an Internet connection, like a home ADSL. In third place, also the infrastructure of PSTN has changed: the dedicated physical telephone line remains such only between the customer premises and the first aggregation point of the provider; from there on, all the voice traffic is usually routed over a

[1] *Public Switched Telephone Network*

country-wide packet-switched network.

On a separated parallel track, another new technology took shape: the so called *peer-to-peer* (*P2P*) *network paradigm*, in opposition to the traditional client/server one.

For the Internet surfer, working the “peer-to-peer way” means being less dependent and under control of a central authority (e.g. avoiding control on file sharing content). For a service provider, working the “peer-to-peer way” means being more scalable and needing less investment on hardware resources, because the providing of the service is distributed among all the peers; in this way users share their resources with the other users.

Skype[78] is the most clear proof of how these two technologies (VoIP and P2P) can be very effective if put together. They distributed a multi-platform P2P-based VoIP application, which required no configuration and had an easy to use GUI[2]. They also offered a very cheap interconnection with the PSTN. Currently, Skype network is reaching peaks of 9 million on-line users at the same time.

Skype, however, is a proprietary and closed solution. For this reason, it is not interoperable with other products and poses some security concerns.

This graduation thesis is going firstly to investigate the current protocols for doing VoIP (chapter 1) and in particular the *Session Initiation Protocol* (*SIP*, in chapter 2).

Then peer-to-peer overlays are examined (chapter 3), devoting particular care to *DHT* (*Distributed Hash Table*) algorithms.

Afterwards, the focus will move on *Network Address Translation* (*NAT*) in chapter 4. NAT is largely employed in *SOHO*[3] networks as well as in big networks installations, because it reduces the need of public IP addresses and is believed to increase network security. However it requires many protocols to be modified to work correctly. NAT traversal techniques will be analyzed, along with the issues that NAT creates for SIP and P2P protocols.

Chapter 5 will analyze how the integration between SIP and P2P is done, and what issues still need to be solved.

[2] Graphical User Interface

[3] Small Office – Home Office

In chapter 6 a solution for effective direct connectivity between SIP UAs will be proposed, in particular extending it to the case of NATted UAs.

In order to perform NAT traversal in a peer-to-peer environment, however, the help of another peer is needed (offering relay service). Therefore, in chapter 7 a methodology for Distributed Relay Service will be proposed. This solution will be also validated by means of statistical considerations and simulation.

In the end, in chapter 8, conclusion about this work will be drawn, and possible future development in this field will be considered.

Additionally, appendix A will discuss the current implementation status.