Summary

From:
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Thesis:
Modernization and optimization of communications via converging voice and data over existing IP network

Annotation:
The term VoIP which stands for Voice over Internet Protocol is a process of digitizing and sending voice telephone signals encapsulated in IP packets over the Internet or a proprietary data network (LAN, WAN/VPN). Deployed over IP-based networks that have proven to be flexible and reliable, the VoIP solutions can save business money by reducing or eliminating the toll charges for long-distance and even local calling. However VoIP is much more than simply a plan to lower a company’s phone bill. There are many so-called “soft” benefits enabled by VoIP, such as increased worker productivity, the ability to collaborate among multiple branch offices, and lower operational expenditures as a result of simplified management schemes. This thesis provides guidelines for planning, designing and implementing a VoIP system for everyone who is looking to make the switch to VoIP or need to make an informed decision about integrated networking. It explains how VoIP works and how it compares to telecommunications technology that was previously considered to be irreplaceable.

Goal:
The goal of the present thesis is creating a profound analysis of the VoIP technology, its reliability, advantages over the legacy phone system, and how it can be deployed. The goal is not only developing a sample solution, but also a detailed examination of the process of migration to converged voice and data network. The same significance has the goal of configuring network devices ensuring the needed security and functionality according to a predefined policy. The goal when creating these configurations is covering the most frequently used data transmission technologies, integration of an existing PBX, as well as implementing the QoS techniques discussed in the thesis.

Structure:
Chapter 1 is a short introduction. It also contains structure description of the thesis and defines its goals.
Chapter 2 describes how the public switched telephone network (PSTN) works, its architecture and its major components.
Chapter 3 covers the advantages of VoIP over the legacy telephone network. Explained are the benefits of the IP telephony applications that can be developed and how they affect the return on investments (ROI)
Chapter 4 is devoted to the standard VoIP technologies. Discussed are the most commonly used codecs, signaling and transport protocols and also some methods of measuring voice quality in IP telephony.

Chapter 5 discusses the quality of services issues. The potential problems when transmitting voice over data networks are analyzed and a solution is given for each of them.

Chapter 6 is focused on the process of creating and implementing VoIP solutions. Among the discussed topics are the infrastructure analysis of an existing IP data network, an overview of the design principles behind the migration strategy, a description of the major phases of a migration and the considerations when choosing network devices.

Chapter 7 is devoted to configuring the network devices for voice transmission. This includes configuring Frame Relay, leased line, PBX and POTS connections, and implementing the QoS techniques discussed in the previous chapters. The command syntax for all of the commands used is explained in detail. Presented are the complete configurations of the network devices for a sample VoIP system.

Chapter 8 is a summary, including the conclusions made and the achieved results.

Results:

The thesis is developed and structured as a complete practical manual for deploying a converged voice and data network. Created are complete sample configurations based on the most common technologies for data transmission, and in compliance with the reviewed best QoS practices, which ensure that all types of network traffic perform up to user expectations. Developed in such manner the present thesis gives the opportunity and could be used in planning, designing and implementing real life VoIP solutions.