LINUX LIVECD VOIP SERVER ADMINISTRATOR MANUAL

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1 INTRODUCTION TO IP TELEPHONY

1.1 INTRODUCTION

This "Introduction to IP Telephony" section explains why companies are converting some or all of their telephone systems from dedicated telephone systems (such as PBX) to more standard IP telephony systems.

These conversions allow for telephone bill cost reduction, increased ability to control telephone services, and the addition of new telephone information services. By upgrading their systems, companies can immediately reduce their telecommunication costs 40% to 80%.

Because IP telephony systems allow the end user and system administrators to setup and disconnect telephone numbers and services, this provides increased control over their telephone features and services. IP telephony is usually based on standard data formats (Internet Protocol). This permits information systems (such as product catalog information) to be more easily linked to the telephone system, thus providing the ability for companies to increase sales through interactive telephone and Internet order processing systems. You will learn that not all voice over data IP telephony systems and services are the same.

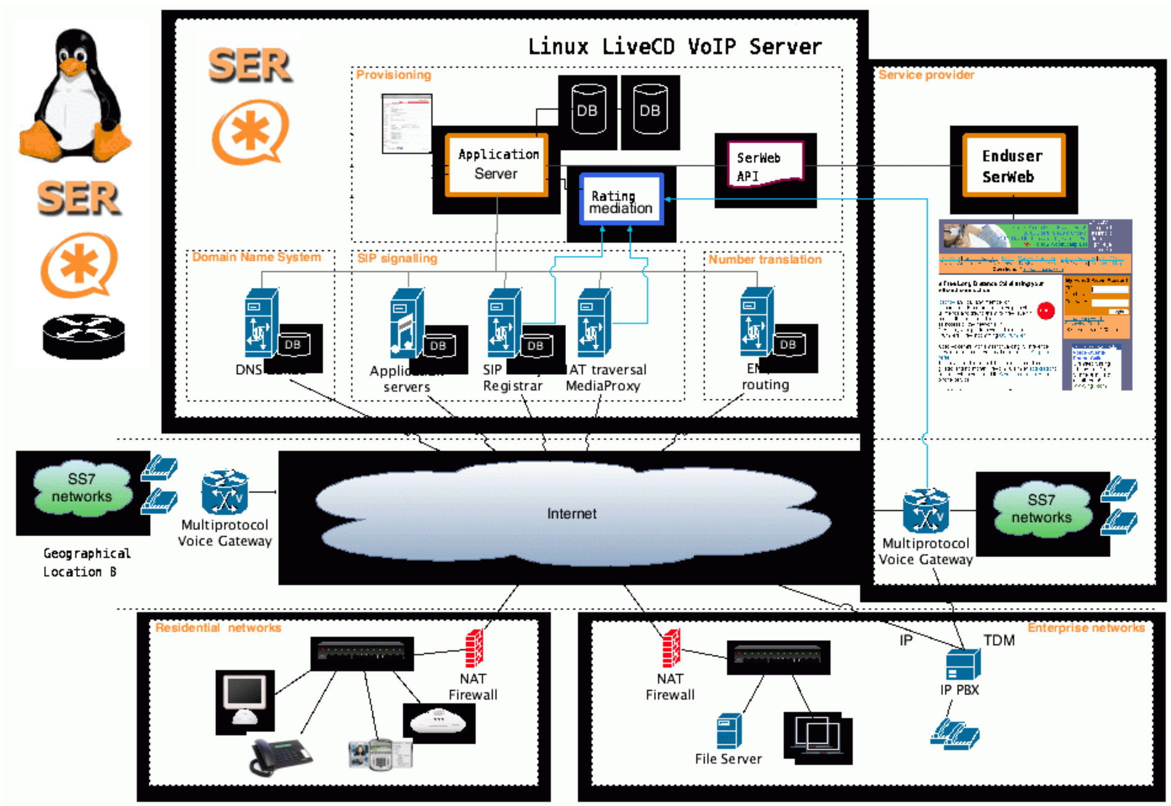
There are cost and quality tradeoffs along with common problem areas and risks. There are many ways these systems can reduce telecommunication costs along with the ability to create new revenue producing services. You will understand how you can get better than telephone toll quality audio, how to maintain or increase system reliability, and new ways to use intelligent telephone systems to increase company revenues. You will learn how employees can keep their phone numbers and existing equipment (using adapters) and call anywhere in the world using IP telephony services. Discover how you can get one (or several) international telephone numbers so your customers can use a local telephone number to call you when you are in another country. You will learn how voice over data telephone service usually allows you to setup new telephone services instantly, display your accounting records and bills in real time, and allow you to integrate information systems (such as sales systems) with your telephone networks.

This section explains the basics of how voice over data telephone service works. This includes how the different ways that voice signals can be converted to data signals (not all of them are the same) and how the conversion process can affect your voice quality. Discussed are the basic processes of sending packets through a data network (such as the Internet) and how the losses of packets (and some do get lost) do not usually affect the voice quality. You will be introduced to the different types of voice over data communication systems that are used for company communication networks. This includes public Internet telephone services providers (ITSPs), IP Centrex service providers, and IP private branch exchange (iPBX) systems.

Also covered are the functional types of IP telephony network equipment such as gateways that are used in voice over data network and some key issues to consider when deploying VoIP systems. This will help you to understand the different types of servers including proxy servers, redirect servers, acess control, provisioning, and policy servers. You will learn about telephone number portability.

Next you will learn about the quality of service (QoS), security, and reliability you can expect from voice over data telephone systems and services. Find out how you can get guaranteed toll quality service with some voice over data telephone systems and why you may experience echoes and audio distortion on others. Learn about how secure your connections are and the different forms of security including the control of physical access, authentication checks, and data encryption. Understand how to maximize the reliability of your calls by understanding the reliability of different parts of the network. Finally, you will discover some of the advanced features and services that are possible with Internet telephone service that is not possible with most traditional telephone systems. This includes unified messaging, ways to connect telephone extension anywhere in the world, how you can setup one (or many) global telephone numbers that will ring to your IP telephone without international charges. Learn how voice over data telephone service allows you to share notes, pictures, and files using a whiteboard while you are talking for interactive web seminars (Webinars). If you are considering converting some of your telephone systems and services be able to use IP telephony services or you just want to know more about the options and advantages of IP telephony services

1.2 NETWORK DIAGRAM



[LINUX LIVECD VOIP SERVER NETWORK DIAGRAM]

1.3 SIP EXPRESS ROUTER (SER) COMPONENT

SIP Express Router (SER) is an industrial-strength, VoIP server based on the Session Initiation Protocol (SIP, RFC3261). It is engineered to power IP telephony infrastructures up to large scale. The server keeps track of users, sets up VoIP sessions, relays instant messages and creates space for new plug-in applications. Its proven interoperability guarantees seamless integration with components from other vendors, eliminating the risk of a single-vendor trap. It has successfully participated in various interoperability tests in which it worked with the products of other leading SIP vendors.

Its performance and robustness allows it to serve millions of users and accommodate needs of very large operators. With a $3000 dual-CPU PC, the SIP Express Router is able to power IP telephony services in an area as large as the Bay Area during peak hours. Even on an IPAQ PDA, the server withstands 150 calls per second (CPS)! The server has been powering our iptel.org free SIP site withstanding heavy daily load that is further increasing with the popularity of Microsoft's Windows Messenger.

The SIP Express Router is extremely configurable to allow the creation of various routing and admission policies as well as setting up new and customized services. Its configurability allows it to serve many roles:

network security barrier, application server, or PSTN gateway guard for example.

1.4 ASTERISK B2BUA COMPONENT

Asterisk B2BUA solution can use IAX2, SIP termination providers. Can do Codec transcoding. Class 5 telephony features. Distributed RTP Proxying. Can provide A-Z wholesale service for multi port gateways, asterisk (SIP or IAX2) or other ser proxys.

1.5 VOIP ARCHITECTURE AND SECURITY

How to assure that your VoIP deployment is secure ?

First we have to accept that security, in general terms, is overhead. It is something we add to the base transport of packet data. As such, security impacts performance, and call quality is one aspect of performance. To achieve total operational support, we have to balance many factors. For some people it is the simple balance of security vs. quality.

Finding balance in the network is much more like balancing the tire on a car. There are many angles and aspects to consider. Take firewalls as an example. When you inspect packets in a firewall, you add latency or delay. We often call this nodal delay. If you think of the firewall as a node in the network, through which traffic must be processed, just inserting a firewall adds delay. Firewalls operate through a rules engine that inpsects each packet and compares it to a set of rules. This takes time, and delays processing.

The same types of delays can be added by intrusion detection systems, antivurs engines and a number of security measures. The trick is to achieve the best possible security without degrading VoIP services.

To achieve this balance, it is important to perform a solid network readiness assessment test. You need to evaluate your requirements, your network, and the ability to meet those requirements. This is all part of the design phase of building your VoIP service. Then you have to test your assumptions about security and call quality to ensure validity. Can your netwokr really support VoIP services without re-design.

Once you have deployed VoIP, you absolutely need to perform some consistent monitoring of network performance to measure ongoing call quality. The security posture of a corporate network changes constantly. New attacks surface, Traffic patterns change. Firewall rules change. And this happens every day. Each of these impacts the call quality your users experience. Effective monitoring of a corporate environment is needed to delivering acceptable call quality.

Perhaps the most important thing to remember is that all delay is cumulative and impacts end-to-end delay. Delay absolutely impacts call quality. So many things we do to strengthen security add delay, that maintaining a balance between call quality and security is vital. That means you need to deply the right tools. tools to monitor quality and performance, and tools to monitor security. It also means that the service delivery team who supports VoIP services will need to work closely with the network security team.

VoIP is unlike email. It's an end-to-end service that requires care and attention to assure appropriate call quality. But, when managed well, it brings values in cost savings and efficiency that far outweigh the labor effort. It really needs to be viewed as a total service