Seminario

VoIP

Comunicaciones de Voz sobre Redes IP

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Introduction

09h45 – 10h30

I will start with something that you are familiar with. To demonstrate this point, I will show you a diagram how telephone exchange is setup in the "old" way.

PICTURE: PBX network diagram showing main office and 2 local offices

An example:

Take Camari, which has is head office in Quito and 2 local offices in Riobamba and Latacunga. They are using the public telephony network to call the local offices or to call from the local offices to the head office. The call volume is high and the calls are frequent; the effect of high volume telephone traffic is an expensive phone bill.

Also, the number of phone lines is limited and could prevent customers reaching the sales department due to the fact that all the incoming lines at the local office are occupied.

Possible solution:

There are 2 options:

1. Order more PSTN lines (if this is possible at all) or
2. add a VoIP solution to existing infrastructure.

PICTURE: Show picture with added VoIP which will be marked with another color.

What is VoIP actually?

VoIP is short of Voice over Internet Protocol. This means that an internet broadband connection established via (A)DSL or a satellite connection will be utilized to transport the voice signal.

VoIP has been designed around 1973, but it’s evolution and acceptance was only recently reached in 2004. This resulted that VoIP grew out to a mature and commercial viable alternative for voice calls.

In order to connect two VoIP devices and to make them "talk", the same protocol must be used. Protocols are rules that every connected device must follow. The most common VoIP protocols at this moment are H.232 and SIP. There are also other less commonly used standards or specific applications like Skype and IAX2.

H.232 is mostly used by telecom carriers, while SIP is used for businesses and home users. IAX2 is a popular protocol widely used on an Open source platform called Asterisk. In this seminar we will discuss what you can do with Asterisk.

As you might know the file formats and compression technology for video are different: *.mpg, *.wmv etc. and as well for audio files (mp3, etc.). VoIP also uses a compressed manner to carry voice information. In order to transfer voice files, you have to squeeze the data in packets which are part of TCP/IP protocol. The technology that allows us to compress, store and send those packets are called CODEC's.

Most known codec are G.711 (aLaw/uLaw), ILBS, G.729 and GSM. The main difference between the codec's are: their bandwidth usage, CPU consumption and error correction capability.

PICTURE: Show table with bandwidth usage

As you can see bandwidth, plays an important role in the choice of which codec to use. When there is enough bandwidth or if VoIP is only used for (fast) internal LAN G.711 is the best choice. But if your bandwidth is limited you should choose the ILBS or GSM codec. I did not mention G.729 due to the fact that this codec is not free.

10h30 – 10h50 - Questions & answer session:

ASTERIKS Basics

Good. All of this is nice, but why we would use such technology? I will try to give you a few clues.

Here are a number of advantages when using the VOIP PBX system:

- Free internal calls, low call costs with many VoIP providers, virtual incoming numbers, and mobility, easy and quick integration into almost any existing PBX.
- Virtual incoming numbers allow you to break long distance barriers and allow virtual call centers on anywhere. For example: you can have customer in Europe, but you do not have to have an office in Europe. With a virtual phone number in Europe, which is terminated over VoIP to your local PBX, you can receive the calls in Ecuador.

More advantages:

- Imagine you have to go on a business trip. Bring your laptop, hard phone or PDA and register as extension on the PBX in your office.
- First of all, you can call your colleagues or make outgoing calls, as if you were in your office.
- Secondly, customers which call to your office can be connected directly to your extension without even knowing that you are not in your office. The flexibility of the system reduces costs and raises availability without losing your mobility.

- Easy and quick implementation within the existing PBX infrastructure can be achieved with inexpensive ATA (Analogue Telephony Adapters) which will convert signals from your existing (mostly) analogue phone lines into VoIP extension.

Disadvantages:

The system also have some disadvantages.

- Dependency on internet connection (reflecting as downtime, bad voice quality, possible high connection costs, etc…)
- Vulnerable in the case of power outages
- The system cannot be used for emergency calls

Asterisks Features

10h50 – 11h30

I have explained you the basics of VoIP and the advantages and disadvantages. Let's see how to use it.

In order to use VoIP you have many choices and solutions. Asterisk/FreePBX is the system that we will discuss in this seminar.

So what is Asterisk?

- ❖ Is Open-Source software written under GPL License, which you can use without any cost.
- ❖ A powerfull Telecommunication system containing many advanced PBX features
  Ot is Telephony software framework meaning that you can program almost any telephony feature required in your situation. Of course it will need some techical knowledge to achieve willing feature.
- ❖ Hybrid connectivity is giving you the possibility to connect to almost any well know telecommunications device: Analog, ISDN, VoIP, Fax, SMS Gateway...
- ❖ It doesn't require expensive hardware, but of course it dependents on how many channels you expect to use. For example: An old Pentium III 900MHz, 256MB is enough up to 3 channels with G.711 codec.

ASTERISK FEATURES

Asterisk has many features but I will mention just few most used.

1. Automated Attendant / Interactive Voice Response (IVR) - An automated system for answering incoming calls and routing them based on the caller's responses to voice prompts. It can replace receptionist and manage higher call load without getting tired.
2. Call Detail Records (CDR) – Provides detailed call reports and usage statistics to show an administrator the activity of the phone system.
3. Call Forward on Busy/No Answer - This feature automatically forwards a call to another extension if the called extension is busy or if the called extension does not answer.
4. Call Queuing - A system that allows inbound callers to sit in a "holding room" listening to music on-hold until the next person is available to speak to them.
5. Call Routing - Based on the phone number that was dialed or the number were the call originated from, a call can be routed to a specified extension, group, queue, etc.
6. Call Transfer - This refers to the ability to transfer an existing call to another extension or even external phone number.
7. Caller-ID - Caller-ID is used to display the phone number and other available information of the user that is calling into the system.
8. Conference Bridging - Asterisk has the ability to create conference calls where various persons can simultaneously attend group meetings.
9. Music On-Hold - Asterisk can play music (files) when callers are on-hold or waiting in a queue.
10. Voicemail - Each user in an Asterisk system can have their extension and voicemail account. The voicemail can be retrieved via their phone, from a remote location, sent via email or accessed via a web browser.

As you can see Asterisk has many features, but in order to be able to fully use these features, one must understand how to configure and program the PBX. To reduce the learning curve and achieve a faster implementation, you can start using Trixbox. Trixbox is a visual interface based on the Asterisk platform.

We have chosen and introduce the Trixbox to you, because it's free, simple to install, easy to use and modular. Most importantly, it can be used for advanced tasks.

We will now show you a real life sample to demonstrate how fast the Trixbox can be installed.

PICTURE: Diagram with Camari network.

The goal of this project was to connect a remote location and be able to call and reduce communication cost to a minimum. Both locations should be able to use the advantages of low cost routing with regards to external calls.

- Burn cd with trixbox installation image
- Boot the computer from Cd and follow the steps. After few restarts your new Trixbox server is ready for use
- Login on console with root and password. You will see the url of the management interface.
- Create SIP extensions on both location. Extensions 100-199 are for Quito and 200-299 for Salinas
- Create IAX2 trunks between 2 servers
- Create outgoing routing
- Enable telephony card to convert analog phone line adding one ZAP trunk
- Set inbound call routing
- Set outgoing call routing enabling all external calls to go via analog lines

You are now ready for basic use of your new digital PBX telephony system. More advanced settings are beyond the scope of this seminar. However, future trainings will be programmed.

Questions?

References

- Trixbox - www.trixbox.org
- “Trixbox made easy” – Barrie Dempster, Kerry Garrison. ISBN 1-904811-93-0
- “Building Telephony Systems with Asterisk” – David Gomillion, Barrie Dempster. ISBN 1-904811-15-9