Asterisk: Open Source VoIP

by

Justin Michel

Final Report Submitted to
the Faculty of the Information Technology Program
in partial Fulfillment of the Requirements for
the Degree of Bachelor of Science
in Information Technology

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4-19-13

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4/16/2013
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Date

Date
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I would like to thank everyone that helped me complete my project because without them I would not been able to complete it and graduate. Through this process I have learned the importance of teamwork and reaching out to people for help is very important. No one has all the answers but everyone has some. Specifically I would like to thank a few people.

Guy Guckenberger from MTCI, Cincinnati, was hard to reach at first but after a few months he became readily available for support. He gave me the insight that I needed to get my project business ready. If I ever had any questions he would always answer them...even if it took him a few weeks to answer.

Mark Shearer from Future Technology Solutions, Indianapolis, has given me some personal tips that have made him successful in his business that took years to learn. Whilst giving me technical insight on how to implement my project, he also gave me insight on what customers might want in a VoIP solution and how to make it the best experience possible.
Abstract

The open source community has always had a strong following because of one main reason—it’s free. Why pay for something you are not sure it will be compatible with your infrastructure or easy to implement. Asterisk started as an open source project for a senior design class and has grown into something huge with many implementations both open-source and commercial. You can start using Asterisk at home and it can eventually be used for a small/branch office, call center, and even at the enterprise level to replace more expensive solutions. Asterisk can be used with several different combinations of hardware and interfaces to customize how you use it. You are not locked into a certain vendor for hardware and can use spare or inexpensive alternatives. You can do everything a normal phone system can do whilst only using a leftover PC, existing or used phones, and only paying for the monthly subscription. An Asterisk based telephony system is a good idea for small to medium to large businesses due to its cost-saving benefits and its flexibility to adapt to many environments.
1. Project Description

1.1 Problem

Voice over IP (VoIP) is quickly becoming a household word because of big products like Vonage, Skype, Google Talk, and MSN Messenger. It has become a cheap or free solution for consumers to collaborate with friends and family. The service is not only for consumers but also businesses because of the recent boosts in call quality and reliability. The reliability of a voice providers based off copper or dedicated PRI lines is unmatched compared to Internet voice providers. On the other hand, the cost savings, increased productivity and flexibility that VoIP can provide is commonly overlooked. Small to medium businesses may have heard of a VoIP solution by names like Cisco, Avaya, and Shoretel but cannot comprehend its value and return on investment. They usually stay with their current provider until it either stops functioning or a really good pitch.

The traditional PBX model is proven to be a reliable phone system model that can withstand almost any disasters including power outages. The downside is that it is expensive, difficult to manage, complicated to upgrade, and has high maintenance costs and proprietary hardware. The dedicated copper lines that run to the company are modular and lacked features that the telephone provider can offer and changes have to be serviced by their technicians.

Asterisk is an open source framework that can be downloaded for free and used by any business. Companies are not locked into any proprietary software or hardware and can use their current voice provider but take advantage of all the features of an in house VoIP system. The traditional PBX has been around for decades and limits companies to
use their hardware, their technicians, and their services, but has the best reliability compared to a VoIP solution. Give your company a chance to increase features and save money by switching to an Asterisk based VoIP system.

1.2 Solution

Small businesses want to be just as productive as the giants and they have options but may not be aware of them. With a free open source Asterisk system, small companies can gain the same benefits of a more expensive VoIP PBX system from the other leaders. There are several companies and solutions from Cisco, Avaya, Shoretel, and even Panasonic that all have a small business solution perfectly suitable for a small business, but sometimes costly, without much room to upgrade, and have proprietary hardware and licensing costs. Asterisk on the other hand is a free, open-source framework that allows small companies can substantially reduce support and hardware costs while also being able to customize it however they want. All you need is a server, a few phones, and their existing networking hardware and infrastructure. There also is no need for a technician to make small changes to the system. With the Web enabled administration of Asterisk, minor changes, adds, moves can be done by anyone qualified to use a Web browser.

2. Design Protocols

Using a small company in Cincinnati as a platform, this project was designed to support five employees with the possibility of up to thirty total users. This company is
tired and frustrated with their current legacy system. They want to upgrade and are looking at various options because their PRI/T1 connection constantly has issues and needs to be rebooted every month or so. Voicemail tends to stop working at some point and the management system is out of date and difficult to manage. Asterisk in this situation, paired with a VoIP provider, is planned to replace this system at some point in the future before the system is deployed this system will serve as a test platform.

2.1 Users

Phone systems that will be discussed here are mostly associated with System Administrators, Network Technicians, Help Desk, and Desktop Support Technicians. The users will have to familiarize themselves with the new hardware which may include servers, fxo/fxs cards, pri cards, analog telephone adapters, POE switches, and analog/IP/software phones. The interface to administer the system is mainly Web-based but there is the occasional event that you may need to telnet or SSH into the server to make changes.

The end user for the most part should not notice a difference. Asterisk allows you to use computer hardware to run the phone system for the entire company and the management is simple enough that you may delegate some management to some users to add or change extensions. Using the correct equipment, you can replicate the same feature sets and phone configurations as you did with a traditional PBX but with a substantial cost savings.
2.2 Solution Details

Asterisk’s main appeals can be summed up as:

- Lower Cost
- Less Maintenance
- More Features
- More Flexibility
- More Mobility

Switching to an Asterisk based solution can give your company everything it currently has and more at a lower cost.

2.2.1 Low Cost

Every business depends on a few key essentials. First is capital followed by a place to conduct business. Right after that would be a communication medium like voice or email so they can do business. Now, if you can saving money while also increasing employee productivity with a free, feature rich, stable Asterisk VoIP PBX system you can use the savings elsewhere like the rent. Asterisk is there to save that money to put towards rent or other expenses while also making it easier to work where you are. It does not take much to get a fully functioning Asterisk server running. It has been tested and implemented on a wide variety of low end devices ranging from a Raspberry Pi, spare workstations or servers, and even virtual machines. None of these servers costs any substantial amount of money, if any. They all can support enough users for a small business, one-hundred or more users, and perform as a reliable voice server. There are no
leases on having a PBX system on site, no monthly cost for licensing the server, no cost per user account, and the cost will be the same or cheaper with a VoIP plan because international calling is free.

2.2.2 Features

Asterisk can provide the services the basic needs that any end user will want. These are the ability to make calls, receive calls, transfer calls, listen to voicemail, and put someone on hold. Asterisk handles these functions easily and very cheaply with many other free features that would normally incur an additional cost at your local telephone company. Some key features Asterisk provides are:

- Voicemail to Email
- Fax to Email
- Auto Attendant (IVR)
- Ring Groups
- Video Conferencing
- Call Queues

Features such as interactive voice response (IVR), call queues, and ring groups are geared towards reducing a user’s workload, or in some cases reducing users. IVR can act as the digital receptionist to direct the call to the right person or be put on hold until the user is ready. This can be very useful for a busy business or a call center to have the incoming calls automatically handled with no user interaction.
2.2.3 Flexibility

Asterisk has the flexibility of being everything you want in a PBX and you can customize it however you want to meet your requirements. You can use it to replace your current PBX, upgrade from another VoIP system, or even integrate it into your existing system. With Asterisk you are not locked into anything. You can choose your hardware--servers, cards, phones. You can choose which distribution or bundles for Asterisk. You can even choose who will support and train your users. If you have multiple sites or offices Asterisk can route the calls between each one just like you were right next to each other. There are several free and paid modules that you can download to add more functionality to your PBX...and if you cannot find one you can request it from the free online community or write one yourself. You can dig deep into the Asterisk code to add, edit, or remove parts that wish. With Asterisk you can have it your way.

2.2.4 Mobility

As the employees of today become more and more mobile, businesses will have to accommodate this with newer technology so they can still collaborate with coworkers, partners, and suppliers. With the abundance of mobile productivity devices like laptops, tablets, and smart phones, users have been accustomed to having whatever information they want whenever they want. Regardless of the new trend for emails, video conferencing, and Twitter, voice has still been a huge preference in communication. Asterisk can bridge the gap between being in the office and being on the road. When an employee is off site, he can still receive calls and video conferences from the main office.
through a remote device. Also Asterisk can capture all your voicemail and faxes and send it via email. Once the message is received at the Asterisk server, it is attached to an email and sent to the user; in some cases it will even be transcribed to text form. Faxes will follow the same procedure. Asterisk gives the mobility that every business wants at an affordable price whether the user is at the office, at home, or on vacation—communication without barriers.

2.2.5 Maintenance

Asterisk’s flexibility with hardware contributes to its low maintenance cost. Without the necessity to be locked into a certain vendor or company for hardware or software, you can assemble your Asterisk solution from several vendors. If you find a good deal on some Cisco phones you can get that or if you prefer to buy phones from Yealink you can use those. Some brands are expensive just because of the name and some lesser known manufacturers can surprise you. The competitors will keep the big guys honest. If everything blows up, replacing or upgrading your Asterisk box is as simple as purchasing the new hardware, reinstall the free operating system, and transfer your most current backups to it.

If there needs to be some users added, changed, or removed anyone in the office can handle it. With a traditional PBX, a technician would have to be called out to change a phone number or add a user. With Asterisk, any minor changes can be completed quickly with the Web Interface anything beyond the interface support is still required.
3. Deliverables

A working implementation of an Asterisk based solution supporting at least five users. It will include two laptops using soft phones, IP phones connected via switches, analog phones connected via voice gateways, and cell phones connected via Wi-Fi. They will demonstrate features such as voicemail, call waiting quizzes, ring groups, hunt groups, static/dynamic conferencing, and interactive voice response (IVR). Calls made through the PSTN will be forwarded to an external SIP/VOIP provider.

A spreadsheet containing what an Asterisk based system would cost for a small company of five users and a quote for a larger company of fifty. The quote will include all the prices of equipment as well as the costs associated with services, training, support, and installation.

4. Project Planning

Asterisk is mainly geared for the smaller companies, so the project will emulate an office. First step is to create a baseline to analyze what hardware the business had. Do they have IP Phones or just analog? Do they have a server available to host Asterisk? Then the budget of the company comes into play. Can they afford new phones? Do they need a new server? Should they gamble with hardware they can handle now or future proof themselves with something more expensive? Lastly, the baseline must be cross-analyzed to the budget to see what makes sense. Once they have decided on the hardware, the test environment has to be created to make sure the call flow works.
At the example office, the required hardware could not be replicated in my test environment due to the lack of funds like a Primary Rate Interface (PRI) line and a patch panel to card to dissect the signals. PRI is the preferred voice service according to every expert in the area, but due to lack of funds and the scope of the project I decided to use a cheaper, less secure and reliable service, called Session Initiation Protocol (SIP). It is not as secure as PRI/Analog because all the voice data goes through the public Internet. Due to it not being a dedicated line, it also is more susceptible to quality issues and service interruptions. Recently the office is choosing to switch to a SIP solution through Cincinnati Bell so the project is much more prevalent now.

At the University we have been taught much about virtualization and its importance for cost savings as well as reliability. I started out running my Asterisk server on a desktop server with a virtual machine and it ran well from there. Then I talked to Mark Shearer and he said that he has never heard of an Asterisk Server ran in a virtual environment in production. The reason is that it is not easy to virtualize some of the hardware required to handle the voice data. The same thing can be said when one prefers to use a hardware RAID controller over a software RAID.

4.1 Hardware

For testing purposes, Asterisk was installed on anything from a custom desktop to laptops and even virtualized hardware. After moving away from virtualization, the first hardware tested was a Raspberry Pi (miniature computer). Testing showed it could not sustain acceptable quality for business use due to the low write and read speeds of the SD card. The next phase after the Raspberry Pi was older desktop and laptop system. These
included an older HP Desktop with a Pentium 4 with 1GB RAM and a backup server on an IBM ThinkPad X41 1.50 GHz Pentium M and 512 GB RAM. These two systems were used as the final deliverables—the HP as the main and the IMB as the backup.

The selection of phones is a Grandstream HandyTone HT503, Cisco 7962G, and Grandstream GXP280. The Cisco phone because it was free and an IP phone but soon regretted choosing it. The trouble with that is it requires a Power over Ethernet (POE) switch so one had to be purchased or use a separate power supply. Due to the lack of Web management, a TFTP server had to be configured to store the configuration files and firmware. The configuration files were deployed but it never correctly registered with the server. Without an active subscription with Cisco, obtaining different firmware to diagnose firmware issues was impossible. The two Grandstream devices, the IP phone and analog adapter, were a breeze to register due to them both having Web management. The two Grandstream devices are being used in the final project.

Voice gateways allow analog phones to directly connect to the Asterisk server by converting the analog signal to digital. Generally they are managed with a Web manager but like the Cisco phone, the Cisco MC-3810-V router lacked Web management and the required firmware files. Digium makes VoIP gateways that plug into the asterisk server with USB and that would be the best alternative. Cisco equipment has never worked easily with Asterisk until recently with SPA-series phones.

4.2 Software
Every Asterisk installation has to be installed on a Linux operating system whether it be Debian or Redhat based. Most installations are bundled with the freeware version of Red Hat also known as CentOS. Despite this, Ubuntu was chosen to virtualize the Asterisk server and clients using a non-graphical virtual manager called Kernel-based Virtual Machine (KVM). KVM is an open-source alternative to more expensive products like ESXi (VMware) and Hyper-V (Microsoft). Virtualbox was also tested in headless mode, meaning no graphical user interface and that was successful as well. Virtualizing the Asterisk server did not meet industry standards and the audio virtualization was not up to par. According to experts in the field, Asterisk on a bare metal installation was the only and best option.

Asterisk can be installed by itself as an application or it can be preconfigured into a Linux distribution. There are a few requirements--operating system, Asterisk source code, and optionally a Web front end. Installing each of the components separately has many more points of failure, so choosing a bundled solution is the easiest option and the most popular. A few examples are AsteriskNOW, PBX-in-a-Flash, Elastix, FreePBX, Trixbox CE, and Switchvox Home Edition. Some of these are made freely open-source by their main commercial products like Digium’s Switchvox and Fonality’s Trixbox Pro. The beauty of these distributions is that one does not need to install an operating system then recompile and install Asterisk. You download the image, burn the image to a CD, boot the server from the disk, then let the disk automate the install.

All of the distributions have tested virtually on a desktop with Virtualbox or VMware Workstation 8, and bare metal on a Raspberry Pi, IBM Laptop, and HP mini desktop. After fumbling around with all these Asterisk distribution, DigAnTel was the
best one. It is actually a repackaged FreePBX distribution, similar to Trixbox, but is compiled and updated by Mark Shearer from Future Technology Solutions in Indianapolis, Indiana. It is a fully automated solution that will install the drivers for your phones as well as other hardware like telephony cards if they are installed to the system.

The advantage of using a VoIP solution is that all the voice is transferred over the wire as data. This means that you are not tied to one physical phone anymore. There were several soft phones tested in the project and they were all free or had free versions. X-lite Soft phone, PhonerLite, and LinPhone are software phones most widely used. X-lite free version is a basic phone that is low on resources and looks as good as a pay to use one. The catch is that it has locked down features. PhonerLite and LinPhone are both fully free phones and have all the capabilities that a phone system like Asterisk has but are not as user friendly. Buttons and tabs all over the place, you can access basically any setting without dropdown menus. It is good for a power user but in the hands it can be broken easily by someone less familiar with server and client settings...but it’s free.

There are even versions of Soft phones for cell phones...Android version 2.3 and on have a SIP client built into them that was tested but not as feature rich or even advertised. There are dedicated apps in the Google Play Store. The most functional ones are the 3CXPhone (an Asterisk competitor) and CSipSimple. They both offer basic options like dialing, transferring and hold but also have features to save battery and mobile data usage.

In the final build, DigAnTel was chosen as the server because it is a custom build from Mark Shearer who installs this for customers on a daily basis and seems to have everything that FreePBX offers and more. Specifically it comes with a nice splash page.
that allows a user to access the management portal, user/voicemail portal, and the
operator panel that lets you get a quick glance at the active phones, conferences, queues,
trunks, and parking lots. The recommended soft phones for users are X-lite and
PhonerLite for the power users in the organization whilst 3CX for the Android and Apple
cell phones and tablets.

4.3 Services

All telephony systems need an external voice provider. This is the one thing most
companies cannot host in their networks. There are several choices to choose from
hosted/cloud Session Initiation Protocol (SIP), Primary Rate Interface (PRI), or analog
lines. SIP and PRI are among the most common but for different reasons. PRI has been
around for decades and is still used for its low cost and reliability. SIP has been popular
for its low cost and flexibility. PRI is a direct line to the telephone company and does
require some equipment installed at the office but has greater than 99% uptime. SIP only
requires the existing Internet connection and can be dialed from anywhere there is
Internet but does not have the same reliability or call quality.

The flexibility of SIP is where it really shines and sometimes can trump PRI in
terms of reliability. SIP lines can be added to a company within a few minutes. One call
to the SIP carrier and one can be created for you, ready to be used. One cannot say the
same for PRI. With a PRI, if you go over your limit of 23 lines, it takes on average 30
days for the provider to have it setup for you. And with that line you are paying for the
full 23 lines. You cannot just add a single line. Also, if your company burns down or the
Internet goes down, simply take the phones (and maybe the server) to another location and business can continue.

These two services can be run without an Asterisk server but what the Asterisk server provides is more control over how calls are handled and several important features. If a call comes into the office building through the PRI without an Asterisk server, it could only be received at the office. But, with an Asterisk server on premise, that call can be redirected from the office through the Internet to a remote extension or phone.

4.4 Budget

Much of the hardware in the project was not used in the end product. Below is a list of what was used in testing and production:

- $35 Raspberry Pi
- $62 Raspberry Pi
- $0 1 Cisco 7962G
- $60 Grandstream GPX280
- $27 Grandstream HandyTone HT503
- $50 CISCO routers (used for analog lines)
- $0 HP Desktop
- $0 CISCO Catalyst 2950 switch
- $0 Server used for testing

The table below compares the cost of the final project versus the actual cost for a small business to support 5 users. There are five phones, one switch with power over Ethernet
(PoE), and one server to host Asterisk and store the voicemail. It does not include any installation costs or maintenance.

<table>
<thead>
<tr>
<th></th>
<th><strong>Actual Cost</strong></th>
<th><strong>Business Cost</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Phones</strong></td>
<td>$60 Grandstream GXP280</td>
<td>$157 x 5 Yealink YEA-SIP-T38G</td>
</tr>
<tr>
<td><strong>Analog Adapters</strong></td>
<td>$27 Grandstream HandyTone HT503</td>
<td>$230 x 1 Grandstream GXW4008</td>
</tr>
<tr>
<td><strong>PoE Switches</strong></td>
<td>$0 used power supply</td>
<td>$500 x 1 HP Procurve 2610</td>
</tr>
<tr>
<td><strong>Server</strong></td>
<td>$0 Server</td>
<td>$700 x 1 Server</td>
</tr>
<tr>
<td><strong>Maintenance</strong></td>
<td>$0</td>
<td>??</td>
</tr>
<tr>
<td><strong>Total</strong></td>
<td><strong>$97</strong></td>
<td><strong>$2230</strong></td>
</tr>
</tbody>
</table>

The cost of the voice provider can be found below. Remember that both SIP and PRI are good choices. PRI is often the preferred choice because of its reliability, but SIP has also proven to be extremely flexible and just as reliable as PRI with the right carrier.

<table>
<thead>
<tr>
<th><strong>SIP</strong></th>
<th><strong>PRI</strong></th>
<th><strong>PSTN</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>$20 x 5 lines</td>
<td>1 PRI with 23 lines</td>
<td>$60 x 5 Lines</td>
</tr>
<tr>
<td>$100</td>
<td><strong>$500-700</strong></td>
<td><strong>$300</strong></td>
</tr>
</tbody>
</table>
4.5. Timeline

5. Proof of Design

5.1 Servers

5.1.1 IBM X41 Laptop

This laptop will host the whole Asterisk phone system.
5.1.2 Compaq
This desktop is the Openfire Server

5.1.3 Complete Setup
This is the way the servers are stacked. The switch interacts between the servers, Internet, and local network and phones.

5.2 SwitchVox UI

Switchvox has a way of making some things easier by coming both the Dialing Rules and Dial Patterns. This is an example of how to setup an Outgoing Call through the Web GUI.
Every Asterisk system so far has had a system status page. This one shows the status of the VoIP providers, SIP and Peer, as well as the active phones and extensions.
What is a PBX system without voicemail? This shows the voice mail inbox for the user at extension 101. Note that they use the same extension to both register the phone and check for email.
5.3 Software Phones

5.3.1 CSIPSimple

This is a demonstration of the software CSIPSimple for the Android. First picture is the dialer and the next image is the accounts on the phone. Note that with a digital phone one can have several extensions.
5.3.2 PhonerLite
PhonerLite is one of the easier to configure desktop soft phones.

5.3.3 X-lite

X-lite phone as seen below is the freeware version of a commercial one provided by CounterPath. It supports basic functions like call, receive, hold and transfer. The paid
version adds features like call recording, auto answer, and video.

5.3.4 LinPhone

This is the only free phone so far that supports video. It has clients on Windows, Mac OS, Android, and IOS (Apple).
5.4 Hardware Phones

5.4.1 CISCO 7962G

This is Cisco’s 7962G IP Phone that can support SIP and SCCP depending on the firmware.
5.4.2 GrandStream GXP 280

This is one of Grandstream’s basic IP Phones. It works well but according to many experts it is only suggested for testing.
5.4.3 Grandstream HandyTone 501

This is the analog phone adapter by GrandStream. It can be bought for about $30 and allows to connect one phone to the network and the PSTN.
6. Testing

There are several key points to test before an PBX system can be setup. For a basic set of testing procedure, in each scenario the call should be able to dial every number and use the functions hold, forward, dial, hang-up, and call voicemail.

6.1 Internal calls

The internal calls are the first extensions to text. The extensions are usually three numbers but it can be anywhere up to four. These calls stay inside the network behind the router and are easy to test. Any extension from should be able to dial any other extension.

6.2 Outbound Calls

The internal calls are one of the simplest to test. Every time I setup the server this is the second thing I test. The outbound dialing rules have to be set to allow calls to the telephone provider. In my case with the SIP provider, I have to send the numbers in the format 1NXXNXXXXXXX, e.g. 513-556-1111, or else the call will fail. So I have to propend some dialing rules on the outbound trunk to add these numbers if I dial a 10- or 7-digit number.
6.3 Inbound Calls to DID

Right after outbound calls are completed, the next phase is to test calls coming into the server. Each Direct-inward Dial (DID) can be routed to an extension, IVR, call groups, queues etc. The DID will be in the format 1 (513) 555-0000.

6.4 Remote Extensions

Remote extensions act just like the internal ones except the voice server is set to the Asterisk server’s outside address. Most times it also has to choose the correct ports, default 5600, to allow SIP to communicate. In the diagram it is represented by Ext 105.

6.5 Ring groups

Rings groups are optional but are common in most companies. They allow one number to into the Asterisk server and can ring multiple phones at the same time. They can be dialed from the internal network or from the outside. This allows a company to save on their phone bill. In the diagram it represented by Sales Group Ext 601.
7. Conclusion

Some companies are still using old or expensive PBX infrastructures by choosing to use the analog PBX or expensive VoIP systems from Cisco or Avaya. Asterisk can help to bridge the gap between a full featured PBX, low cost voice service, excellent reliability and failover, all while saving a ton of money. The low hardware requirements, cost savings with a VoIP/SIP provider, flexible and reliable transfer between digital and analog voice server providers, and scalability due to the open source nature of Asterisk make it a great solution for any company’s PBX.
8. Definitions

Dial Rules - used for adding numbers to, or subtracting numbers from the number being sent to the trunk

Dialing Patterns - used to specify what numbers are allowed to go out via that route. If a dialed number does not match the pattern it is not sent through the trunk

FXO - Foreign eXchange Office

FXS - Foreign eXchange Service

IVR - Interactive Voice Response

Openfire - free, open-source instant messaging server that runs on Linux/Mac/Windows

PBX - Personal Branch eXchange

POE - Power over Ethernet

Port - virtual connection point that allows software to communicate without interfering with each other

PRI - Primary Rate Interface

Raspberry Pi - low-cost, basic computer contained on a single circuit board

SIP - Session Initiation Protocol

Trunk - it carries the call to a VoIP service provider (VSP)

VoIP - Voice over Internet Protocol (IP)

VSP - VoIP service provider
9. References


<http://www.youtube.com/watch?v=xn33BdpF5y4>.

"SIP Trunking: The savings are there but the transition is complex". 04 Apr. 2013.