Getting Started with UC

Discover the concept behind UC and how you can make money from it from it with as little effort as possible. Our stories are backed up by real-life experiences. Read all about saving time with easy management, saving money with automation, and profiting from the popularity of social networking.

Hardware infrastructure, setup and configuration

The **server** is the central component of a hardware infrastructure designed to offer Unified Communications. It's a physical / virtual machine with an **Internet connection** to which all computers and phone terminals are connected. **Linux** (CentOS and RHEL are also supported by VoipNow) and your **PBX** should be installed on this server. The hardware specifications vary depending on network capacity and expected performance.

In-house

If you already own an adequate server, you may install an IP PBX and you'll be ready to provide Unified Communications.

Hosted

Your server resides in a remote data center. It's important that you choose a data center that's geographically close to most of your customers to provide redundant, low-latency connectivity.

In the cloud

Some organizations offer infrastructure as a service. For instance, you can rent a server from Amazon Web Services.

You need to make sure your hardware configuration meets your network's needs. You can decide on the performance of your server taking into consideration the maximum number of concurrent calls you expect to be made. The general formula is that for every concurrent call (with a G. 711 uLaw codec), your server will use approximately the resources listed below.

Each concurrent call = 33MHz CPU + 8 Mb of RAM memory + 86 Kb of your network's bandwidth

How extensions affect the server

The number of simultaneous calls impact the performance. For example, if your server will provide service to residential clients, most of the extensions will probably be Phone Terminals, which do not require many resources. But if you resell towards businesses, things like IVR, Queues and other costly extensions will require more powerful hardware.

How the server handles PSTN

SIP trunks can route calls to/from PSTN networks on their own, no need for a physical gateway. You might be required by law to have a PSTN connection.

What about codecs

Codecs are designed to enhance VoIP calling sound quality. However, these codecs influence your server’s or your network’s performance.

For instance, G.711 uLaw and G.711 aLaw do not compress packages upon sending them. This means that they will improve your call quality, but they will consume bandwidth. On the other hand, if you use G729 – which is one of the most used codecs – the packages will be compressed and your call quality will be amplified, but this action translates into more CPU workload.

This is why it's recommended that you calculate your system to have at least 30% resources free most of the time, so that you won't have to worry about peaks.

Set up the software

VoipNow is a platform that includes an IP PBX, billing engine, a web-based administration interface and multi-tenant management.

At the core of the VoipNow system lays the **IP PBX,** or Internet Protocol Private Branch Exchange, which is a private telephony network used within a company. It is a software application that implements the basic functions of a traditional hardware PBX as well as next-generation digital services.

Basically, it **acts like an automatic switchboard** that connects users. In addition, it offers new services that traditional telephony lacks due to technology limitations or high cost.

Am I safe?

A major concern for VoIP users is the system's security. Internet telephony may seem liable to hacking at first. In fact, all internet solutions are. The risks of using VoIP do not exceed those that have already been accepted by businesses integrating the Internet with all their activities.

Compromised user accounts can be easily detected with basic monitoring solutions and damage is usually restricted to a few international calls placed on the user's behalf, without affecting the system's integrity.

Call your colleague

All you need is a phone. While most features of Unified Communications solely require a computer with internet access, VoIP calls are a little more pretentious. You will need a **phone terminal** to be able to make and receive calls, and there's a sea of options to choose from these days.

One of these options is to install a **softphone** on your computer. There are various softphones available online and some are free of charge. By installing this software, you can use your computer to make and receive calls as long as the program is running and provided that you have headphones or speakers and a microphone.

For a more familiar experience, you can purchase **IP phones** or even use **regular phones**. The main function of the phone terminal is to transform analog audio signal into digital and vice-versa. IP phones are built to do this, but regular phones can also be connected to a VoIP network using a VoIP adapter.

SIP carriers show you the world

After setting up your internal hardware infrastructure, you can make VoIP calls, but only inside your local network. Connecting to the public telephone network requires the services of one or several **SIP carriers** .

A SIP carrier is a VoIP carrier whose equipment supports the SIP (**Session Initiation Protocol**) standard. SIP is currently the most popular VoIP protocol on the Internet, used to connect millions of networks and devices.

SIP carriers operate an infrastructure that can **route** and **connect** the calls to/from from your system to phone numbers on other infrastructures (IP networks or PSTN). They own media gateway machines that act as an interface between different types of networks by converting between the different transmission and coding technologies.

DIDs help you hear and be heard

With all the characteristics described above, you will be able to establish VoIP calls. However, in order to **receive calls**, you will need DID or **Dire ct Inward Calling**. This feature can be offered by your SIP provider.

The telephone company provides the customer with one or several trunk lines for connection to the customer's PBX, allocates a range of telephone numbers to this line (or group of lines), and forwards all calls to such numbers via the trunk.

To reach users with VoIP phones, DID numbers are assigned to a communications gateway connected by a trunk to the public switched telephone network (PSTN) and the VoIP network. Calls originating in the VoIP network will appear to users on the PSTN as originating from one of the assigned DID numbers.

Automation makes it all easier

In addition to a visual interface, VoipNow also offers **billing engine** operations.

With a regular IP PBX, you most likely would have to export call records and then process them either manually or using another software product. However, this is not time-effective and the reports may not be precise.

VoipNow's charging plans automate this operation and the call history based reporting is always **exact**.

To Sum Up

1. Make sure you set up an **IP network** and configure compatible **servers** suitable for the infrastructure you have planed.

2. Choose the most suitable **codecs**.

3. Set up your software. With **VoipNow**, installation is done from the browser interface and you need not worry about the **IP PBX**.

4. Set up the devices: with *VoIP, you can install **softphones**, **IP Phones** or regular phones.

5. Expand your network with the help of **SIP carriers** and interconnect with other communication networks.

6. Make yourself available for calling with **Direct Inward Calling**, which can be provided by your SIP carrier.