Basics of VoIP

Version 1.1

July 26, 2006

AdvOSS.com

sales@AdvOSS.com
support@AdvOSS.com

Phone: +1 647 259 8660

Table of Contents

Executive Summary

This whitepaper precisely introduces the emerging technology, known as VoIP (Voice over IP). A VoIP call is initiated between two points, both or any one of these points can be PSTN (Public Switched Telephone Network) lines or PCs (Personal Computers). The whitepaper completely describes all these scenarios in detail with a network diagram for better understanding.

Whitepaper provides brief introduction of “Soft Phone”, “Gateway”, “Gatekeeper”, “RADIUS (Remote Authentication for Dial in User Service)” and “CDRs (Call Detail Records)”. It also discusses different business models for VoIP Providers.

Introduction

VoIP stands for Voice over Internet Protocol. IP is the protocol over which the whole Internet is running. It can be assumed that anyone who can get online to Internet is already running IP.

When voice is carried over an IP network (Internet), the whole process is called VoIP. Since Internet is lot cheaper than the conventional telephony (TDM) process, the voice traffic is quickly converting to IP from the old TDM circuits.

Technical Details

A voice call is initiated between two points. One or both of these points can be normal telephone (PSTN) lines. One or both of them can be Computers called PCs as well. Following figure illustrates VoIP process:

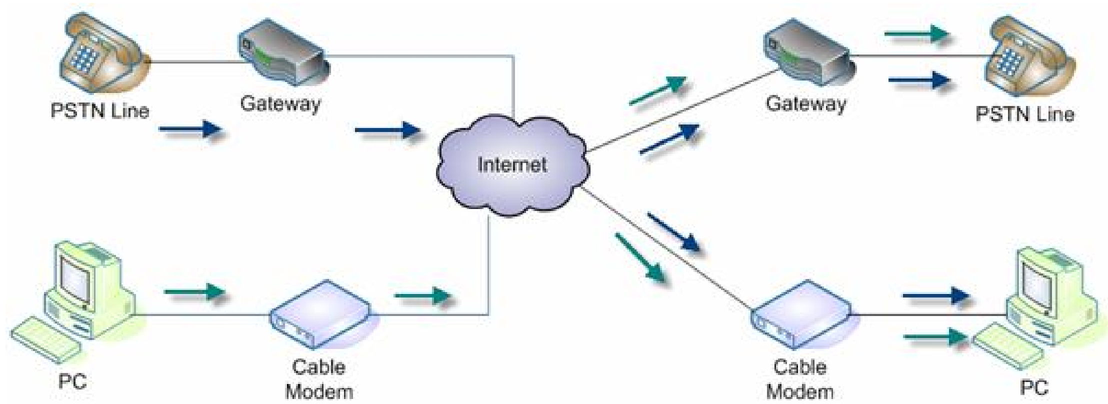

Figure 01: VoIP Process

If both the ends are normal phones, it is called a Phone to Phone call. If the originating end (caller party) is using a PC and the receiving end is a PSTN line, it is called a PC to Phone call. Typical examples are services like Net2Phone and DialPad etc.

Rarely does it happen that both the ends are using a PC in which case it is referred to as a PC to PC call and even rarely it happens that a user is using a Phone to call a PC in which case it is referred as a Phone to PC call e.g. service by Vonage.com.

If one or both of the legs are on conventional PSTN lines, a gateway is required to convert calls from internet to PSTN or from PSTN to Internet.
The two end points on Internet (PCs or gateways whatever the case may be) need to agree on one signaling and one voice exchange protocol. The most common signaling protocol is H.323 and another is called SIP.

The protocol used for voice exchange is called a codec and commonly used codecs are G.729A or G.723.1. These are called compression protocols. The default protocol that does not use compression is called G.711.

If the user is using a PC to originate or receive a phone call, he will need software called soft phone or a dialer. This software has the signaling protocol as well as voice codec built in. Unlike other software which only runs on the servers, this software runs on each and every client's computer. Following figure depicts a dialer:

Figure 02: A Dialer (Product of AdvOSS)

Each voice protocol requires a specific amount of Bandwidth to communicate with the other end. G.711 requires 64Kbits/second and is therefore not used in VoIP. G.729A requires 8Kbits/sec for voice traffic and including overheads require a total of 16Kbits/sec. G.723.1 requires 5.6Kbits/sec and with overheads require a total of 16Kbits/sec.

From now on we will assume that the hardware and software being used uses the G.723.1 protocol.

This means that one voice channel requires 16kbits/second. If you intend to support say 30 simultaneous voice channels, a total of 480kbits/sec will be required. After allowances it is assumed that a 512kbits/second link is enough to carry 30 simultaneous voice channels on G.723.1 or G.729A.

PSTN Lines

PSTN lines come in two common standards. Normal phone lines coming in your home are called analog lines. They have one dedicated phone number and can be used to originate or receive one voice call at a time.

There are other types of lines called Digital lines. The connectors look more or less the same as the normal analog phone cable but a digital line has multiple PSTN lines built in them. One type is called a T1 line. It has 24 PSTN lines in it and runs on 1.544 Mbits/second. The other type is called an E1 line and has 30 PSTN lines in it and runs on 2 Mbits / Second. T1 lines are mostly used in USA and Japan and E1 lines are used elsewhere in the world.

Small VoIP operators usually work on 4 to 16 analog lines while medium to large VoIP operators work on E1 or T1 lines.

Gateway

Gateways are classified as supporting analog or digital lines. A typical low end gateway will support 4 analog lines and costs usually in the US$100 to US$500 range. A typical gateway supports 4 digital lines (96 to 120 PSTN lines) and costs around US$10,000 to US$20,000 range. Following figure depicts a gateway:

Figure 03: A Gateway

Once a call is initiated someone has to account for the authentication of that call and see if the calling party has enough credit to make the call. Typically software called a RADIUS Server is used for this purpose.

Gatekeeper

Software called Gatekeeper is also required to find out the destination of the call and route the call to the proper gateway that can handle the delivery of that call. Some gateway hardware by some manufacturers has most of the gatekeeper functions built into them. In that case an external gatekeeper may not be required.

Gatekeeper or gateways generate all the detail records of all the calls made. These are called CDRs (call detail records) and are used by billing server for the billing. CDRs are either generated online through the RADIUS protocol and are billed there and then. Alternately CDRs can be .csv text files which can be processed by the billing server later.

Business Models

A VoIP provider usually has two business models:

1. Origination
2. Termination

1. Origination

Origination is again of two common types.

i. Phone based origination
ii. PC based origination

- In phone based origination, the VoIP provider arranges for gateways with a local number that can be used by customers to initiate calls. Please see the "Basics of Phone to Phone VoIP" white paper for more details.

- In PC based origination, the VoIP provider arranges for users to use their computers to initiate voice calls. Please see the "Basics of PC to Phone VoIP" white paper for more details.

2. Termination

In this business the VoIP providers arrange for a gateway that takes calls off the Internet and delivers to PSTN lines. Please see the "Basics of VoIP termination business" white paper for more details.

Summary

VoIP stands for Voice over Internet Protocol. A voice call is initiated between two points i.e. both or one of these points can be conventional telephone lines/Computers called PCs. If one or both of the legs are on conventional PSTN lines, a gateway is required to convert calls from internet to PSTN or from PSTN to Internet.

The two end points on Internet need to agree on one signaling and one voice exchange protocol, most common signaling protocol is H.323 and another is called SIP.

Once a call is initiated RADIUS Server is responsible for the authentication of that call and sees if the calling party has enough credit to make the call.

Software called Gatekeeper is also required to find out the destination of the call and route the call to the proper gateway that can handle the delivery of that call.

Gatekeeper or gateways generate CDRs (Call Detail Records) for each call.

A VoIP provider usually has two business models:

- Origination
- Termination

Contact Information

In case of any ambiguity regarding the concept, explained in the whitepaper, please feel free to contact us at support@AdvOSS.com or please, visit http://www.AdvOSS.com/voip_contact.html

For further information please, visit www.AdvOSS.com

We welcome your suggestions

Thank You for reading this whitepaper. We will be pleased to receive your response and suggestions.