Beginners Guide to Voice over IP (VoIP)
VoIP Introduction

Internet telephony refers to communications services—voice, fax, SMS, and/or voice-messaging applications—that are transported via the internet, rather than the public switched telephone network (PSTN). VoIP uses internet protocol data packets to transfer voice, fax, and other data over the shared network, thereby eliminating toll charges, which is why they are cheaper than calls over PSTN. Constant technical innovation and upgrades are being made in this sector to make VoIP quality comparable to the traditional phone system.

This traditional PSTN system works by setting up a dedicated channel between two points for the duration of the call. This system is based on copper wires carrying analog voice data over the dedicated circuits. VoIP, in contrast to PSTN, uses what is called packet-switched telephony. Using this system, the voice information travels to its destination in countless individual network packets across the Internet. These packets need to be assembled and then converted to voice. VoIP systems can be connected to the traditional PSTN system.

In layman’s terms VoIP is the transmission of voice over the digital network. The steps involved in originating a VoIP telephone call are as follows:

- Signaling and media channel setup
- Digitization of the analog voice signal
- Encoding
- Packetization
- Transmission as Internet Protocol (IP) packets over a packet-switched network

On the receiving end, the above steps take place in the reverse order:

- Receiving the IP packets
- Decoding of the packets
- Digital-to-analog conversion which reproduces the original voice stream

**Codecs**

To enhance the security of the packets being transmitted and to prevent any unauthorized illegal access, they need to be encoded. Codecs encode and decode both ends of the conversation to allow the conversation to be sent and received across the network.
Different codecs have different bandwidth requirements and different characteristics that can impact
network performance. Some of the popular codecs and their characteristics are defined as follows:

G.711 Codec - It uses compression schemes and hence requires more bandwidth.

G.729 Codec - It uses lossy compression which reduces the amount of bandwidth required, but can impact the
voice quality.

Once the codec has its payload ready, it’s up to another protocol, to transfer data to its target recipient. So for
applications like audio and music streaming, a protocol like RTP (real-time transport protocol) can be used
where acknowledgement of the protocol is not a key factor.

Protocol

The word protocol comes from the Greek word protocollon which means a leaf of paper glued to a manuscript
volume that describes the contents. In other words, a protocol is a set of rules or procedures that needs to be
followed to allow an orderly communication. Webopedia defines it as “An agreed-upon format for transmitting
data between two devices.” The protocol should determine the following:

- Type of error checking that needs to be used
- Data compression method that needs to be used
- How the sending device will indicate that it has finished sending a message
- How the receiving device will indicate that it has received a message

There are a variety of protocols that one can choose. The protocol can be either hardware-based or software-
based. The implementation decision will depend upon a number of factors including the following:

- Advantages and disadvantages of the protocol being used
- Complexity involved in its usage
- Speed, quality, and reliability

Defining VOIP Protocols

There are a number of protocols that are employed in order to provide for VoIP communication services. They
can be implemented in using both the proprietary and open protocols and standards. In order to be able to
communicate using a VoIP system, there are two types of protocol that must be used.

1. **Signaling protocol**

- controls and manages the call
- includes elements such as call set up, clear down, and call forwarding
- examples of such protocol are H.323, SIP, and skinny
2. **Data exchange protocol**

- manages the data exchange for the VoIP traffic
- handles both audio and video (e.g. RTP)

In this report, we will focus on those important protocols that are widely used and accepted in the industry:

**H.323**

- Packet-based multimedia communications systems
- Distributed architecture for creating multimedia applications, including VoIP
- Widely deployed and is the oldest of the call setup protocols
- Robust and flexible
- Lots of handshakes and data exchanges for each function it performs

**Media Gateway Control Protocol (MGCP)**

- Defined as IETF RFC 2705
- Centralized architecture for creating multimedia applications, including VoIP
- Uses UDP on port 2427
- Lets a call server control a VoIP gateway to the PSTN. Hence, the bulk of the call control intelligence and routing information resides in the call server, instead of the gateway because of this call server controlling the gateway

**Session Initiated Protocol (SIP)**

- Defined as IETF RFC 2543.
- Defined as a distributed architecture for creating multimedia applications, including VoIP
- Lightweight protocol that accomplishes almost everything as H.323 with much less overhead
- SIP client interfaces are shipped with Microsoft Windows XP
- Designed for applications that need real-time performance to send data in one direction with no acknowledgments.
- Examples of companies offering SIP phones are Cisco and Avaya

**Real-time Transport Protocol (RTP)**

- Used to report on the performance of a particular RTP transport session
- Delivers information such as the number of packets transmitted and received, the round-trip delay, jitter delay, etc. that is typically used to measure QoS (Quality of Service) in the IP network
• This is an “unreliable” protocol built on top of the UDP protocol that does not guarantee delivery of packets, but has little overhead
• Widely used for streaming audio and video data.
• Designed for applications that need real-time performance to send data in one direction with no acknowledgments

Session Description Protocol (SDP)

• Intended for describing multimedia communication sessions for the purposes of session announcement, session invitation, and parameter negotiation
• Does not deliver media itself but is used for negotiation between end points of media type, format and all its associated properties

H.248 (Megaco)

• An ITU Recommendation that defines “Gateway Control Protocol”
• It is also known as The Media Gateway Control Protocol (Megaco) which is the result of a joint effort between the IETF and the ITU-T Study Group 16

Skinny Client Control Protocol (SCCP)

• Along with these standardized call setup protocols, vendors have provided their own proprietary protocols. E.g. Cisco has provided Skinny Client Control Protocol (SCCP)
• It is a simple, lightweight call setup protocol for Cisco devices and passes messages using TCP and port 2000

Who defines the Protocol

• IETF (Internet Engineering Task Force) - This community of engineers that standardizes the protocols that define how the Internet and Internet Protocols work. Refer to the following link for further details: http://www.ietf.org/
• ITU (International Telecommunications Union) - An international organization within the United Nations System where governments and the private sector coordinate global telecom networks and services.
Service quality

For VoIP to be a realistic replacement for standard public switched telephone network (PSTN) telephony services, customers need to receive the same quality of voice transmission they receive with basic telephone services—meaning consistently high-quality voice transmissions. VoIP is extremely dependent and sensitive to bandwidth availability and delay caused in delivery of packets. The major challenge with VoIP implementation is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due interference from other lower priority traffic. Hence the following things may be considered when defining the Quality expected from VoIP:

- Latency: delay for packet delivery
- Jitter: variations in delay of packet delivery
- Packet Loss: too much traffic in the network can cause the network to drop packets

Voice Myths

Cisco provides an excellent distinction between the network and what myths we have regarding them in their “Understanding Voice over IP Protocols”.

<table>
<thead>
<tr>
<th>Myths</th>
<th>Facts</th>
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<tbody>
<tr>
<td>Networks can only be built one way</td>
<td>VoIP allows centralized or distributed architectures</td>
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<tr>
<td>Networks will only use one protocol</td>
<td>H.323, SIP, MGCP and H.248/Megaco will all be present in VoIP networks</td>
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<tr>
<td>All networks will converge</td>
<td>The networks will converge to IP</td>
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The Author is a Project Manager in a leading multinational organization. In his current role, he leads software projects aimed at driving defined business value. A multi-skilled professional with over 12+ years of total experience and an excellent track record of managing complex projects across various technologies (SAP BI, MSTR, SharePoint and .NET) and domain. Able to manage stakeholder expectations and willing to take full responsibility for the delivery of projects objectives. Has an excellent track record of determining goals, time frames, constraints, risks, staffing requirements and procedures for accomplishing projects. Has played diverse roles and has a passion in teaching, trekking and reading books.

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Dorset Connects is a leader in providing IT strategies and consulting to small and mid-sized organizations throughout the Philadelphia Metropolitan area. In business for seventeen years, Dorset Connects prides itself on providing clients with a truly “Hassle-Free” IT experience. From network assessments and support, to full time staffing assignments or business continuity planning, Dorset Connects is your IT Buddy!