9.2.2 VoIP Facts

Voice over IP (VoIP) is a protocol optimized for the transmission of voice data (telephone calls) through a packet-switched IP network. VoIP routes phone calls through an IP network, including the internet. VoIP solutions can integrate with the public switched telephone network (PSTN) to allow VoIP customers to make and receive external calls.

In this lesson, you will learn about:

- How VoIP works
- VoIP servers
- VoIP endpoints
- VoIP gateways
- VoIP protocols
- VoIP codecs
- QoS, latency and jitter

How VoIP Works

The following sequence occurs during a phone call between two VoIP phones within the same organization.

1. A phone number is entered on a VoIP phone.
2. The VoIP phone contacts a VoIP server, also known as an IP PBX.
3. Using the phone number, the VoIP server determines the IP address of both phones.
    - If needed, the VoIP server can contact another VoIP server to determine the distant VoIP phone's IP address.
4. The VoIP server helps establish the connection between the two VoIP phones.
5. When a user speaks, the VoIP phone digitally encodes the sound and adds the digital data to IP packets.
6. The IP packets are carried by the network to the distant VoIP phone.
7. The distant VoIP phone unwraps and decodes the data.
8. The VoIP phone converts this data to audio that can be understood by the distant user.

VoIP Servers

The heart of any VoIP system is the VoIP server, also known as a VoIP PBX or IP-PBX.

- The main role of a VoIP server is to provide a switching mechanism to connect internal calls between VoIP endpoints.
- VoIP servers also provide a switching mechanism to connect calls between internal VoIP endpoints and shared external PSTN trunk lines.
- VoIP servers connect to the network like other servers.
- VoIP servers can also connect to PSTN trunk lines.
- VoIP servers maintain user accounts.
    - Accounts map phone extension numbers to VoIP endpoints.
    - Accounts have usernames and passwords to provide authentication and security.
- VoIP servers provide many of the functions of a PBX.
    - Call waiting
    - Call transfer
    - Conferencing
    - Voice mail
- Good VoIP servers have internal POTS ports that interface with older devices, such as fax machines that require an analog signal.

There are alternatives to purchasing a dedicated VoIP server.

- Purchase VoIP services from a provider located in the internet.
    - The provider may require the purchase of their VoIP phones.
    - Any disruption to your internet service will also disrupt your cloud-based phone system.
- Use VoIP server software.
    - Add this software to an existing server or run the software in a virtual machine.
    - Additional hardware may be required, especially when connecting to PSTN trunk lines for external calls.

A VoIP trunk can be used as an alternative to PSTN truck lines.

- VoIP trunk lines are purchased from a provider located in the internet.
- VoIP trunk lines come with phone numbers that your VoIP server will use to accept incoming calls and send outgoing calls.

VoIP Endpoints

The clients of a VoIP server are called VoIP endpoints. They include:

<table>
<tr><th>VoIP Endpoint Type</th><th>Description</th></tr>
<tr><td>Hard Phones</td><td>
The standard VoIP phone is a hard phone. Hard phones:

Have the look and feel of a traditional phone with a base and a handset.
Are really computers that are built to look and operate like phones.
Are connected to a network, just like any other computer.
Can be configured using a web browser connected to their web page interface.

An analog phone can be converted to a VoIP hard phone using a converter box.

The converter box is, essentially, a hard phone.
The analog phone becomes the phone’s handset.

For convenience, electrical power for the VoIP hard phone is supplied through twisted pair Ethernet cabling. Most VoIP phones are manufactured according to one of two IEEE 802.3 power over Ethernet (PoE) standards. Be sure that your PoE equipment matches the requirements of your VoIP phone.

PoE follows the original 802.3af standard and provides15.4 watts of DC power.
PoE+ (or PoE Plus) follows the updated 802.3at standard and provides 25.5 watts of DC power for Type 2 devices.

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<tr><td>Soft Phones</td><td>
A soft phone:

Is a software application that is installed on a computing device like a computer or a handheld device.
Allows access to the VoIP server for real time audio communication.

If the software is installed on a computer, the computer's microphone and speakers act like a soft phone handset.
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Whether they are hard phones or soft phones, VoIP endpoints must be configured with a username and password that matches an account on the VoIP server. Otherwise, they will not be allowed to communicate with the VoIP server.

VoIP Gateways

A VoIP gateway converts voice and fax calls between the PSTN and your IP network in real time.

- VoIP gateways are generally smaller, less costly, and more specialized than VoIP servers.
- VoIP gateways can be placed in geographically separated branch offices to save long-distance PSTN charges.
  - A call from the main office switched through the PSTN to a distant external phone would incur long distance charges.
  - A call from the main office traveling over an existing WAN to a VoIP gateway and then switched though the local PSTN to the same distant external phone would incur only local PSTN charges.

VoIP Protocols

VoIP uses protocols that reside on top of transport layer protocols like TCP, UDP, and SCTP to assemble and distribute VoIP data throughout your network.

- Major manufactures of VoIP systems use proprietary protocols.
  - One manufacturer's VoIP server may not work with another manufacturer’s VoIP phones.
- Session initiation protocol (SIP) is an open source VoIP protocol.
  - Most major manufactures make SIP-compatible phones and phone systems.
  - Smaller manufacturers almost exclusively use SIP.

VoIP Codecs

A special algorithm called a *codec* compresses VoIP data to reduce bandwidth consumption.

- A codec compresses the data on the sending end and decompresses the data on the receiving end.
- A codec determines the sound quality of the VoIP call and the amount of bandwidth that it will require.
  - A codec that preserves sound quality may require a large amount of bandwidth to transmit.
  - A codec that requires less bandwidth may have poor sound quality.
- A VoIP system's default codec should be adequate for small originations.
- When VoIP data consumes large portions of bandwidth, consider changing the codec.
  - Free open source codecs may be a good choice.
  - When purchasing a codec, find one that conserves sound quality while lowering the required bandwidth.
  - A good codec will give high quality sound and consume about 4.5 Kbps of bandwidth.

QoS, Latency, and Jitter

QoS (Quality of Service) can assign priority depending on the type of network traffic.

- Network devices like switches and routers can examine the type of service bits and precedence bits in the header of an IP packet to determine the type of traffic.
- QoS settings on network devices can be configured to give priority to VoIP traffic.

Network latency is how long it takes for a packet of data to get from one point to another.

- Too much latency causes VoIP callers to talk over each other.
- Experts advise that 250 milliseconds is the maximum level of latency that is acceptable in VoIP systems.
- Latency between 75 to 150 milliseconds is conducive to acceptable quality for VoIP conversations.
- Network latency is most important when a VoIP server is hosted in the cloud because of the added internet latency.

Jitter is a variation in the delay or latency of received packets.

- Latency going up and down during a call can cause unusual sound effects (minor pauses, jumps, choppiness).
- When troubleshooting jitter, first check QoS settings.
- VoIP endpoints can be configured with jitter buffers.
  - Jitter buffers add delays and can introduce latency issues.
  - Identify and correct the sources of jitter before considering jitter buffers.
