Telephony VoIP IP Interview Questions And Answers Guide.

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Telephony VoIP IP Job Interview Preparation Guide.

Question # 1
Explain the IPV4 and IPv6 address bit?

Answer:-
yes IPV6 is advanced version of IPV4, the address of IPV^ is 128 bits with extend able memory and that of IPV$ is 32 bits.

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Question # 2
What is IP addresses range A,B,C,D,E?

Answer:-
0.0.0.0 126.255.255.255--class-A
127.0.0.0 127.255.255.255--LoopBack IP address.
128.0.0.0 191.255.255.255--class-B
192.0.0.0 223.255.255.255--class-C
224.0.0.0 239.255.255.255--class-D
240.0.0.0 255.255.255.255--class-E

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Question # 3
Tell me voip test setup that you have for testing?

Answer:-
Press and release the mute button, than press the following number on dialpad 8378# (TEST), This start self test.

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Question # 4
Explain url?

Answer:-
URL stands for : Uniform Resource Locator.
To access information over internet using web browser URL is used as a address to access particular web site.
URL is the conversion of IP to human understandable form.
Users give the name of website in the browser which is later on converted to IP address by DNS server. And the request page is send to users browser.

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Question # 5
Do you know in voip telephone witch part will convert data anolog to digital, digital to analog?

Answer:-
A codec (Coder/Decoder) converts analog signals to a digital bitstream, and another identical codec at the far end of the communication converts the digital bitstream back into an analog signal.
In the VoIP world, codec’s are used to encode voice for transmission across IP networks.
Codec’s for VoIP use are also referred to as vocoders,
for "voice encoders". Codecs generally provide a compression capability to save network bandwidth. Some codecs also support silence suppression, where silence is not encoded or transmitted. Regards

**Question # 6**

Explain How does VOIP work?

**Answer:-**
The basic principle of VoIP is very simple. It’s the same technology you have probably used already to listen to music over the Internet. Voice sounds are picked up by a microphone and digitized by the sound card. The sounds are then converted to a compressed form, compact enough to be sent in real time over the Internet, using a software driver called a codec. The term codec is short for "encoder/decoder". The sounds are encoded at the sending end, sent over the Internet and then decoded at the receiving end, where they are played back over the speakers. The only requirements are a connection between the two computers of an adequate speed, and matching codecs at each end.

To be usable, a VoIP system also needs a method for establishing and managing a connection, for example, calling the other computer, finding out if they accept the call, and closing the connection when a user hangs up. Because VoIP allows two way communication, and even conference calls, it’s a lot more complicated than simple audio streaming. How calls are managed is the area in which VoIP systems fundamentally differ, and two VoIP users must be using the same system (or compatible ones) in order to be able to call each other.

Because most Internet users don’t have a permanent Internet address (IP address, a number like 212.44.88.17 that uniquely identifies that computer, at that moment), VoIP systems don’t generally work by calling another computer directly although that may be an option for those who do have a permanent address. Instead, each user of the service registers with an intermediate server, which maintains a record of their IP address all the time they are connected. An example of a VoIP application that works this way is PicoPhone. The small size of the PicoPhone application file (about 64Kb, barely larger than Windows Notepad) demonstrates clearly that the basic principles of VoIP are not complicated to implement.

Another reason for using an intermediate server is that it eases the problem of getting VoIP to work through the firewalls that everyone uses these days. Many firewalls block any data from the Internet that is not sent in response to a specific request. This makes it impossible to call another computer directly. Because the called computer did not request any data from the caller, the call request would be blocked. By establishing a connection with a server, the VoIP software opens a channel of communication through which other computers can call it. Communication may continue using the server, or information may be passed via the server that allows the two computers to open a direct connection between them and continue using that.

**Question # 7**

What do internet telephony, packet telephony, IP telephony and converged network means?

**Answer:-**
The first thing all mean the same thing. Which is using IP (Internet protocol) for voice services. Some voice networks are only packet-switched and have no access outside of their own VoIP network. Most VoIP networks have a Gateway that connects to a circuit-switched external network which gives them access to external calling. One of the gateways responsibilities is to convert G.711 Circuit-switched media (typically a T1 provided by a telco company) to the 7.723 Packet-switched media that will traverse the companies VoIP network. A device called a gatekeeper will then convert the IP address (used by H.323 protocol) to a standard telephone number (E.164 address) that can be used for external calling.

A converged network is a network that passes both Voice and
Data over the same set of devices. Converged networks generally implement QoS (Quality of service) on all active network devices to ensure the VoIP has priority over standard data because of its more rigid demands.

**Question # 8**

Explain VOIP?

**Answer:-**

VOIP means voice over internet protocol. In which the voice is send through internet (or any IP network). For that the analog voice is converted into digital data and use appropriate CODEC (for bandwidth saving) and send through IP network. At the receiving end the digital data is converted back into analog voice.

**Question # 9**

Explain What are the advantages to VoIP?

**Answer:-**

The big advantage is VoIP may save you money depending on how much you are currently spending for local and long-distance calls. What you will need to do is get the total cost the phone company is charging and compare it against a VoIP plan that interests you. With most plans, you get free calls within the U.S. and Canada for a low flat rate. International calls usually have very low rates with no connection fees. For both residential customers and businesses that make a lot of long distance and international calls, the savings can be several hundred dollars a year.

Another advantage is with the features available with VoIP. Features such as caller ID, call waiting, call forwarding, 3 way conferencing and voice mail are usually included at no extra cost. With the phone company, these services are usually extra.

In addition, you can make free phone calls anywhere there is a high speed Internet connection available. That means you can be in another state or even in another country and make calls as if you were back at your home or business. You will just need to bring your phone adapter along with you and possibly a phone in case one is not available.

**Question # 10**

Tell me What equipment is needed for VoIP?

**Answer:-**

The main requirement is a broadband Internet connection such as DSL or cable. Any other equipment such as a telephone adapter or microphone usually comes with the VoIP service provider.

**Question # 11**

Tell me How does VoIP work?

**Answer:-**

To understand how VoIP works, you will be taken through the process of voice transmission from one end to the other. The process starts with a person talking into the mouthpiece on one end of a VoIP call. This analog voice signal must first be sampled and digitized. Voice sampling is usually done 8,000 times per second (8KHz). In order to reduce bandwidth, a voice CODEC is used. A voice CODEC is a compression/decompression algorithm that is optimized for the voice frequency range. The bit stream uncompressed is 64Kbps. By using an available CODEC, the bit stream can be reduced to 8Kbps or less.

In order for the compressed voice data to be sent over the Internet, it must go through a process called packetization. This is a packet consisting of a small sample of the voice data (usually 10-30 milliseconds). While being routed through the Internet, these packets can get delayed or even lost. This can cause degradation in voice quality. Simply put, there are various mechanisms in
place to compensate for these problems and help smooth out the audio.
Once all the packets arrive on the listening end of the call, they must be reassembled to their original state. The packets are decompressed and converted from a digital to analog voice signal.

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