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# What is VOIP?

VOIP is also known as Voice Over Internet Protocol. It is a phone service over the internet because it can transfer voice and multimedia content over the Internet Protocols (IP) networks. VOIP historically was referred to using Internet Protocols to connect private branch exchanges (PBXs). VOIP is used by a group of technologies used to deliver voice over either the internet or local area networks or VOIP endpoints. If we have a usable Internet connection, we are able to get a phone service delivered through the internet connection instead of using traditional phone calls from the local phone service provider such as Maxis, Digi or etc.

People use VOIP instead of the local phone service provider because of the lower price than traditional phone calls or sometimes VOIP offers free services for user too. Even though VOIP offer cheaper price than traditional phone calls, VOIP sometimes does not offer 911 service or other common phone services. But, VOIP’s new digital nature allows new feature that are either expensive or impossible to be implement by traditional phone service provider such as voicemail, call diverts, integration with PC that allow us to call a number just by using a web browser or email and many other more. VOIP can be used to call any phone in the world and it does not matter what network or equipment the person calling is using.

VOIP is available to many personal computers and other devices that have internet access nowadays. Calls and Short Message Service (SMS) can be sent over mobile data or Wireless Fidelity (Wi-Fi). VOIP allows modern communication devices which includes smart-phones, tables and etc to be consolidated using a single unified communications system (UC).

The steps and principle that used to make VOIP calls are very similar to traditional calls. It involves signaling, channel setup, digitization of analog voice signals and encoding. VOIP transmit voice using a packet-switching network. Packet-switching is method of grouping data and then transmitted over a digital network into packets. VOIP transport media streams using special media delivery protocols that will encode both video and audio with audio and video codecs.

# Overview of the system architecture that applied voice-over-IP protocol.

­­­­The system that applied voice-over-IP protocol include Skype, WhatsApp, Facebook Messenger etc. These systems’ call features are implemented with VoIP. In VoIP, there are two parts that need to be solved. The first is the setting up for the calling, for example the call signaling and the second is the carrying of the actual voice traffic back and forth between the caller and the receiver. The meaning of call signaling is the establishing of the connection between the caller and the receiver. This is something like the Transmission Control Protocol (TCP) three-way handshake. The Session Initiation Protocol (SIP) and H.323 are some well-known call signaling protocols.

SIP is the IETF’s standard for establishing VoIP connection, it is also an application layer control protocol for creating, modifying and terminating sessions with one or more participants. The architecture of SIP is something similar to HTTP, requests are generated by client and then sent to the server, after processing the request then a response is sent back to the client. A request and the response for that request make a transaction. SIP itself provides reliability and does not depend on TCP for reliability. SIP is depending on the Session Description Protocol (SDP) to carry out the negotiation for codec identification, it supports session description that allow user to agree on the compatible media types. SIP provides user mobility support through proxying and redirecting requests to get the most accurate user’s location.

As for H.323, it is the ITU-T’s (International Telecommunications Union) standard that vendors should comply while providing Voice-over-IP service. The reason why this is the standard of ITU-T is that it provides technical requirement for media communicate over LAN’s while assuming that no Quality of Service (QoS) is being provided by LANs, it was originally developed for multimedia conferencing in LANs but later extended to cover VoIP. The H.323 system has four logical components, Terminals, Gateways, Gatekeepers and Multipoint Control Units (MCUs). In these components, terminals, gateways, gatekeepers are also known as the endpoint. Terminals are the endpoints of LAN client that provide real time and two ways communication. Gateways is an endpoint same as terminal, but it connects H.323 terminals on the IP address and another H.323 gateway, they act like a “translate” between transmission. Gatekeepers, the most important component in H.323 system, it is a “manager” that acts as the central point for all calls within a zone and gives out its services to the registered endpoints. A zone is the registered endpoint and the aggregate of the gatekeeper. The multipoint control units is also and endpoint on the network, but it provides capability for more than three terminals and gateways to participate.

VoIP apps such as WhatsApp and Skype using public internet as their base of operation. These apps use their own proprietary protocols in call signaling. For reference, WhatsApp used to have a custom version of the Extensible Messaging and Presence Protocol (XMPP), but they have change to signaling protocol now. Most of the apps also use some form of RTP/UDP to carry the voice packets. The last known speech codec used by Skype is called SILK, while WhatsApp and Facebook Messenger use Opus, a high versatile and variant of SILK. Most of the apps used encryption to protect the communication and this increase the difficulty to actual probe into the protocols in use.

However, apps that works over the internet must solve some additional challenges, which is how will calls be set up between user and any location in the world. Most of the time, user connect to the internet via a private network such as Wi-Fi or mobile network (3G/4G) which means that Network Address Translation (NAT) is probably in use. A NAT is the process where a network device, usually a firewall, assigns a public address to a computer inside a private network via the Internet. A communication over the internet requires a public IP address. In this case, a central respiratory that stores the mapping between username or their phone number to the public IP address which user was seen is needed and the list needs to be constantly updated as mobile users move around. If it is a peer-to-peer model, the caller will reach out to one of the server of the app and ask for the public IP address of receiver, the app will check database and send back information to caller and caller may use the information to open a session to the receiver.

From the information that receive from a research paper, WhatsApp app is using TURN (Traversal Using Relay NAT) method for their system architecture. A TURN server is a network entity in charge of relaying media in VoIP related protocol, it serves as a relay point in the protocol that have media flow through it, via Quora. Each of the peer establishes a connection to the TURN server and the server relays the voice packets between the peers. This method is uses to dealt with the problem that might happen, for example, when the calling party tries to initiate a connection to the receiver and a firewall blocks that connection or the NAT on the peer’s side does not allow a connection to be made reliably to the peer.

For Skype application, the device will open a connection to the app’s STUN (Session Traversal Utilities for NAT) server, the server will then reply the device with the IP address and port in which the connection came on. In most time, the STUN server will also reply with the public IP or port of other peer, the calling party can use the information retrieve from STUN server to send a message directly to the receiver or set up the call.

The VoIP technology in the communication application is the wave of the future as it combines numbers of technology within it. Technology will keep on improve as times move on, therefore the VoIP communicate system will become more efficient and cost effective way to takeover current phone calls system.

# How does Voice-over-IP work?

# The advantages and disadvantages of the technology.

# Current issue regarding this technology?

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