WebRTC: Web Real Time Communication

Webrtc is a complex technology now used in the telecommunications industry for testing interactive communication capabilities of voice, video, chat into a web browser. With sudden change in ways how people work due to the pandemic, virtual communications has now become a necessity with billions of people now connecting via video calls.. Companies are building up video, chat applications based on webrtc providing them better audio and video quality.

Introduction:

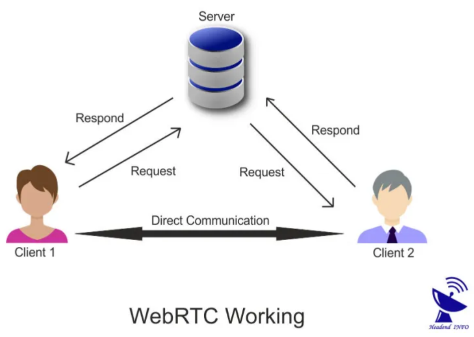
WebRTC is an HTML 5 specification that can be used to add real time media communication between browser and devices. Webrtc enables voice and video communication to work inside the web pages. It offers communication natively from a web browser.

To understand it better lets go a few years back when had to install additional plugins like Adobe Flash Player to play any video. A pop up used to flash up asking to install Adobe Flash Player when we wanted to play a video. WebRTC now eliminates the need for the installation of any third party plugins. All you need is to have a browser that supports webrtc which modern browser like Chrome, Safari, Edge do now.

Lets break WebRTC a bit

- Web RTC or Real time communication, doing communication in real time. Well we can say webrtc is a technology and not a solution. It main focus is to send the data as fast as possible in order to have a low latency.
- Webrtc is free to use, it comes as an open source project which is embedded in the browsers but you will you will be able to adopt it for your own needs.
- Webrtc is not just limited to browsers it is also available for mobile applications, the source code is easily portable into mobile apps.

Communication in webrtc:



When it comes to communicating peer to peer, a connection is established between two peers via a signaling server.

Connection establishment Process

1. Client 1 sends a request to the server to connect to client 2.
2. Client 2 responds to server to send back a response to client 1.
3. Once request and response from both sides are established the two peers are now connected and will be able to communicate directly without ther server.
4. The server here is being used to establish connections between client 1 and client 2 and once the connection is established the two peers will be able to communicate directly.

Webrtc Uses cases :

1. Webrtc is being used in numerous industries one of the major use case is seen in contact centers communications since the pandemic all contact centers have moved to working from home so ther is an increased need to provide network security, remote assistance to client/agent.
2. Webrtc sees a major use in telehealth , online education, online fitness classes , coaching, conducting personalised one to one sessions.
3. Unified Communication-voice and video calling, 1:1 or group sessions.
4. Low latency broadcasting, live streaming, sports game or auction

Webrtc Codecs

Webrtc codec is a computer program that encodes and decodes a data stream or signal.

Purpose of codecs

Voice and video codecs is to compress and decompress the media that needs to be sent over the network.

Lossy Compression: not maintaining the match as to what goes in and what comes out, mainly seen in audio and video streaming.

Lossy compression something that is not being perceived by the human eye or ear.

Lossless Compression: these are codecs, whatever input they see at encoder is same at the decoder end, no loss along the way.

Note:

Codecs basically remove everything which we do not need to hear, since audio and video hold a lot of data which needs to be sent over the network, so in order to avoid any network wastage codecs will remove everything and anything which our ears or eyes dont notice.

Lossy compression is like a virtual dial (regulator) which you could use to reduce or increase the loss of data. This flexibility in compression level setting is used to manage the bitrate.

This page covers the very basics of webrtc, what it is , purpose behind which applications deploy it, but the learning curve of webrtc is quite steep and covers extensive topics of signaling , media processing to name a few.