**WEB RTC Tutorial's :**

References :

* Muaz Khan
* Sam Dutton
* Rob Manson

UDP (connection less) is used for audio video calls cause of its low overhead and ability to send broadcast messages to multiple destinations.

**STUN : Simple Traversal of UDP through NAT**

1. Useful for connecting a device which is behind a NAT or firewall.
2. Will not work if Symmetric NAT routers are used.
3. Always tres STUN first, If it fails uses TURN.

**ICE : Interactive Connectivity Establishment**

1. ICE server is a media relay server which is used to set up the media session.
2. It's a framework which allows web browsers to connect to peers.

**TURN: Traversal Using Relays around NAT**

1. Main disadvantage is it's cost of use. There is a huge bandwidth usage when transferring a HD video stream.
2. STUN is faster than TURN when peer's are behind the same NAT, i.e in a LAN.

**SDP : Session Description Protocol**

1. Describing the multimedia content of the connection, i.e the resolution, formats, codecs and encryption.

**Signalling Server**

WebRTC need's a signalling server to :

1. For the clients to exchange the meta data to co-ordinate communication. : **This is called *Signalling***
2. To cope with NAT's and firewalls.

**SCTP : Stream Control Transmission Protocol**

**What is Signalling?**

Signaling is the process of coordinating communication. In order for a WebRTC application to set up a 'call', its clients need to exchange information:

* Session control messages used to open or close communication.
* Error messages.
* Media metadata such as codecs and codec settings, bandwidth and media types.
* Key data, used to establish secure connections.
* Network data, such as a host's IP address and port as seen by the outside world.