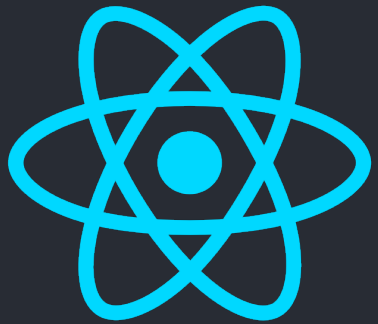
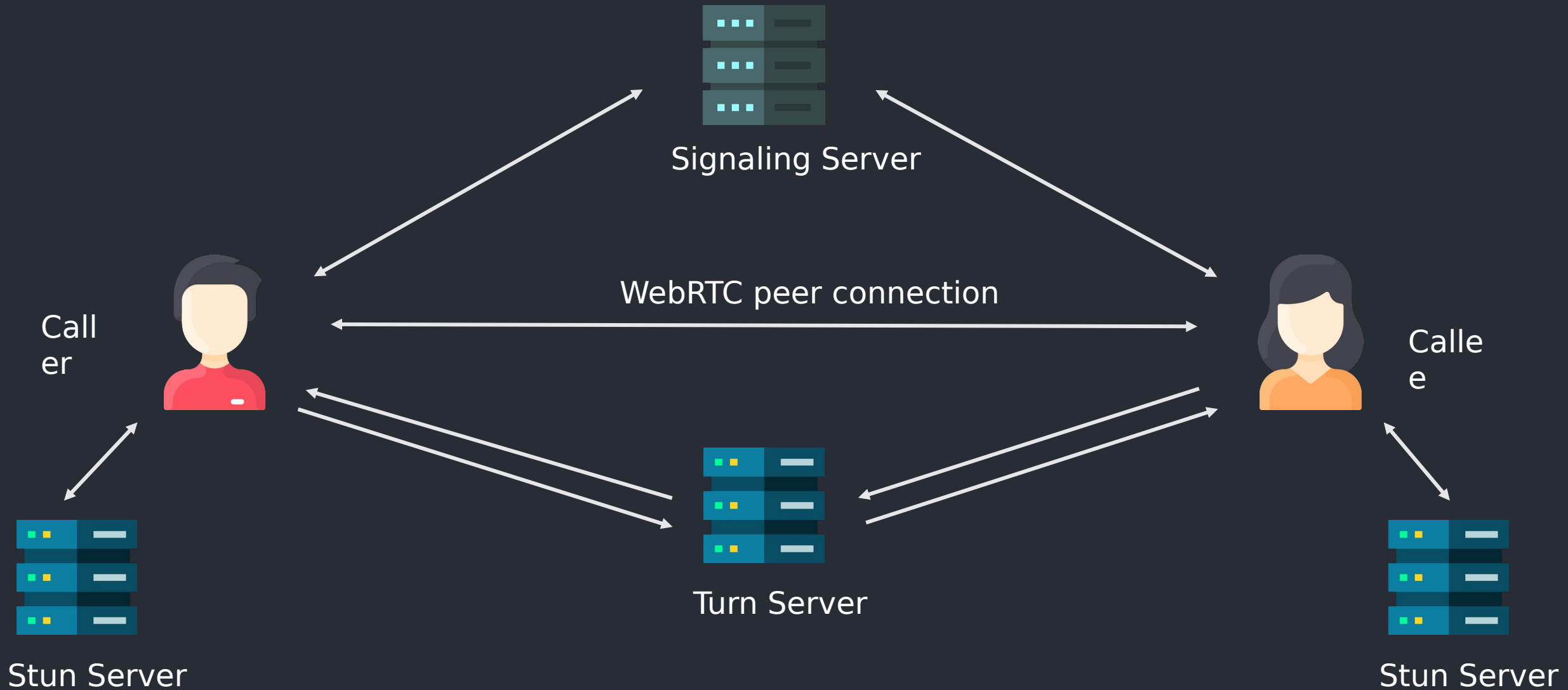


WebRTC



How WebRTC is working ?



What is WebRTC ?

WebRTC (Web Real Time Communication) is an open source project that enables peer-to-peer communication between browsers or the applications (mobile). In other words, WebRTC allows you to exchange any kind of media through the web (such as video, audio and data) without any required plugin or framework.

What is SDP(session description protocol) ?

- The Session Description Protocol is a format to describing multimedia communication sessions for the purposes of session announcement and session invitation
- It does not deliver the media data but is used for negotiation between peers of various audio and video codecs, network topologies, and other device information

Exchanging SDP

- The caller calls **`RTCPeerConnection.createOffer()`** to create an offer with SDP
- The caller calls **`RTCPeerConnection.setLocalDescription()`** to set that offer as the *local description*
- The caller uses the signaling server to transmit the offer to the intended receiver of the call
- The recipient receives the offer and calls **`RTCPeerConnection.setRemoteDescription()`** to record it as the *remote description*

Exchanging SDP

- The recipient then creates an answer by calling `RTCPeerConnection.createAnswer()`
- The recipient calls `RTCPeerConnection.setLocalDescription()`, passing in the created answer, to set the answer as its *local description*
- The recipient uses the signaling server to send the answer to the caller
- The caller receives the answer
- The caller calls `RTCPeerConnection.setRemoteDescription()` to set the answer as the *remote description*

ICE candidates

- As well as exchanging information about the media (discussed above in Offer/Answer and SDP), peers must exchange information about the network connection. This is known as an **ICE candidate** and details the available methods the peer is able to communicate (directly or through a TURN server). Typically, each peer will propose its best candidates first, making their way down the line toward their worse candidates. Ideally, candidates are UDP (since it's faster, and media streams are able to recover from interruptions relatively easily), but the ICE standard does allow TCP candidates as well.

Exchanging ICE candidates

- Offer is created by the caller and set as *local description (SDP)*
- Caller asks STUN server to generate ICE candidates
- ICE candidates are received from STUN server and after setting local and remote description they can be exchanged using signaling server
- Callee also ask STUN server to generate ICE candidates
- Both sides calls *RTCPeerConnection.addIceCandidate()*, when candidate will come through the signaling server

STUN Server

- **STUN (Session Traversal Utilities for NAT)** that allows clients to discover their public IP address and the type of NAT they are behind. This information is used to establish the media connection.
- In 15-20% cases STUN server will fail and to establish connection between the peers we will need TURN server

TURN server

- TURN server is Traversal Using Relay NAT, and it is a protocol for relaying network traffic
- TURN server will be used if STUN server will fail
- TURN server will be used as an assist to establish connection between the peers
- TURN Servers are not public because of the costs which they can generate because of the traffic which is going through them

