




# Videoconference system for E-learning platform based on WebRTC protocol



# About presentation

-  WebRTC – Definition and usage
-  System architecture
-  Centralization of media streams



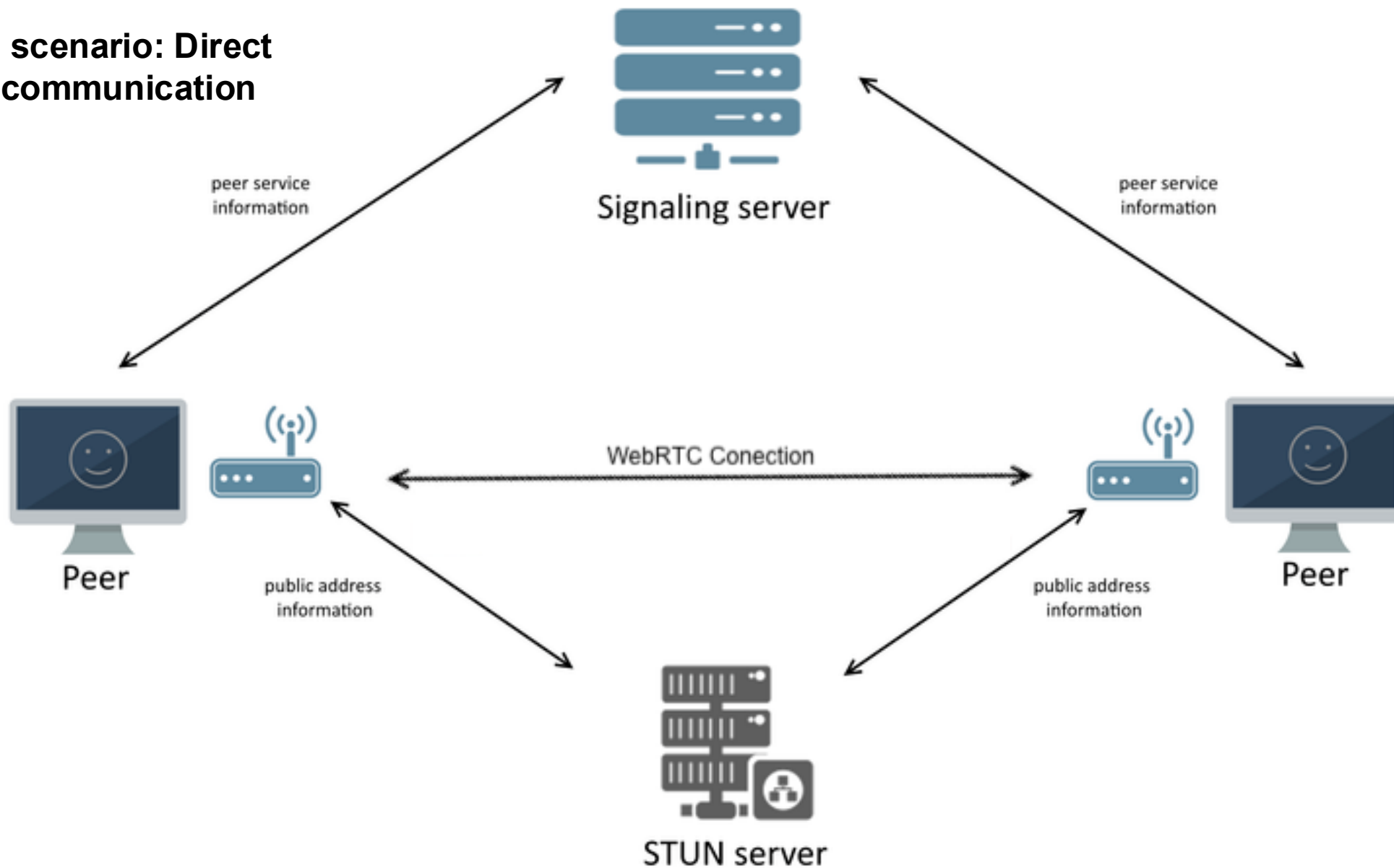
## WebRTC: Definition






WebRTC (Web Real Time Communication) is an open source technology and set of protocols that enables real-time communication – including audio, video and arbitrary data directly between browsers and devices using peer-to-peer (P2P) connections

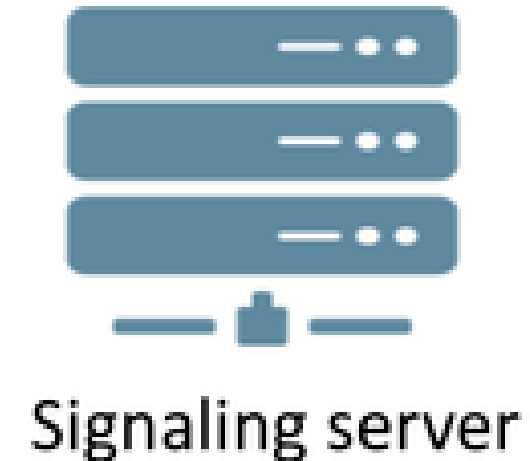


## Ideal WebRTC scenario: Direct Peer-to-Peer communication

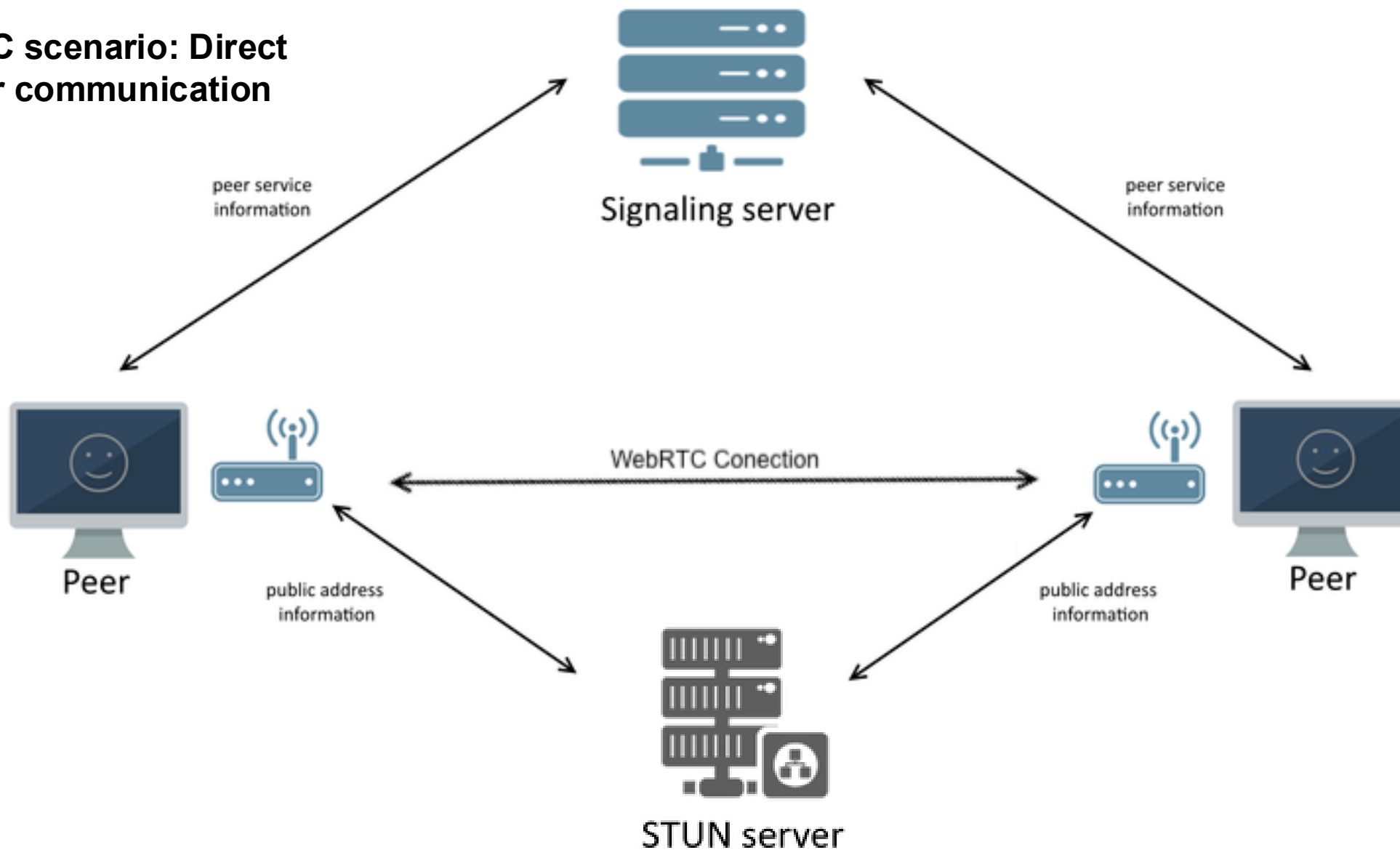


## Signaling Server




-  It's purpose is to exchange connection-related information between 2 peers
-  It is commonly built on top of WebSocket technology to enable low-latency communication
-  The exchanged connection data includes:  
the SDP offer/answer and ICE candidates

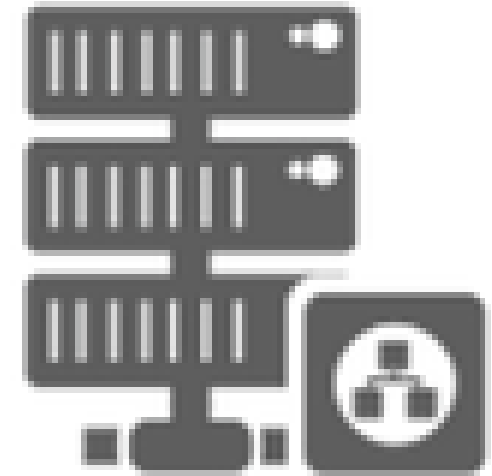


## Ideal WebRTC scenario: Direct Peer-to-Peer communication



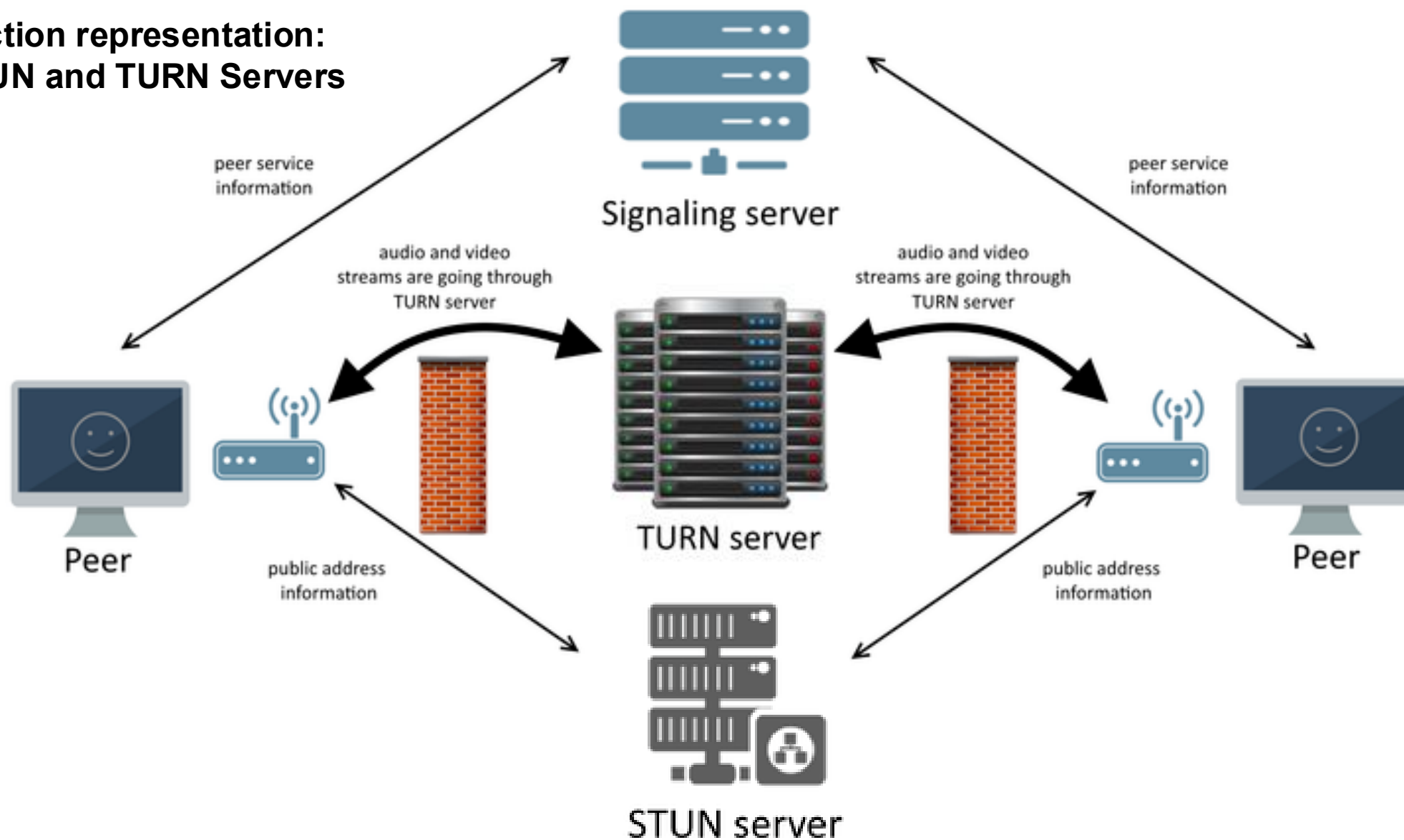
## STUN: Session Traversal Utilities for Nat

-  It's purpose is to determine the public IP addresses of the peers involved in the communication
-  The server is required only during the initial connection setup between peers
-  Beside public IP, it also delivers information about the type of NAT configuration the device is behind






STUN server

## WebRTC connection representation: Scenario with STUN and TURN Servers





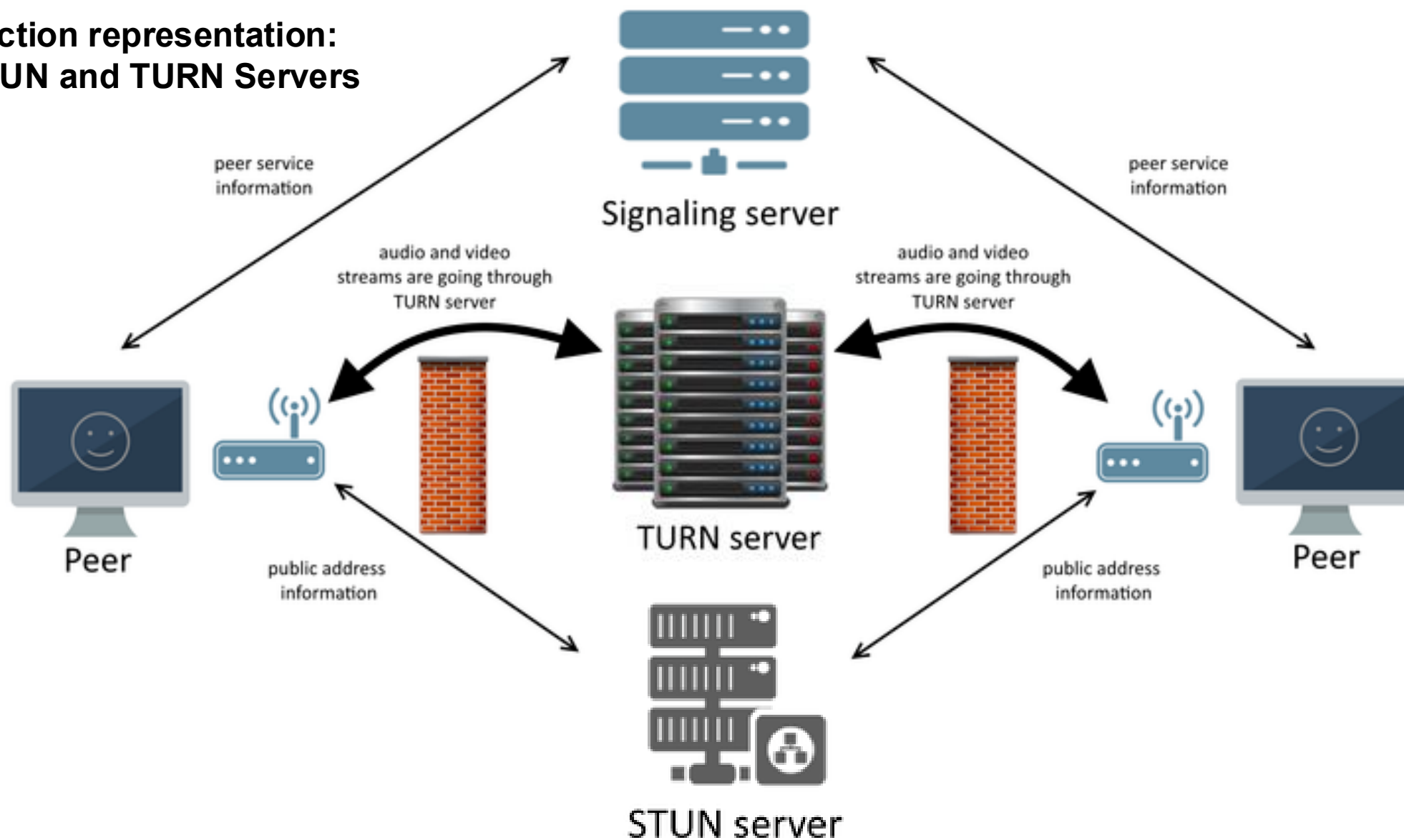
## TURN: Traversal Using Relays around NAT

-  The TURN server's role is to facilitate and intermediate data relaying between participants
-  TURN is used as a fallback mechanism when direct P2P connection is impossible
-  Built to handle various network conditions, it enables adaptive media streaming over both TCP/IP and UDP, ensuring flexibility and reliability

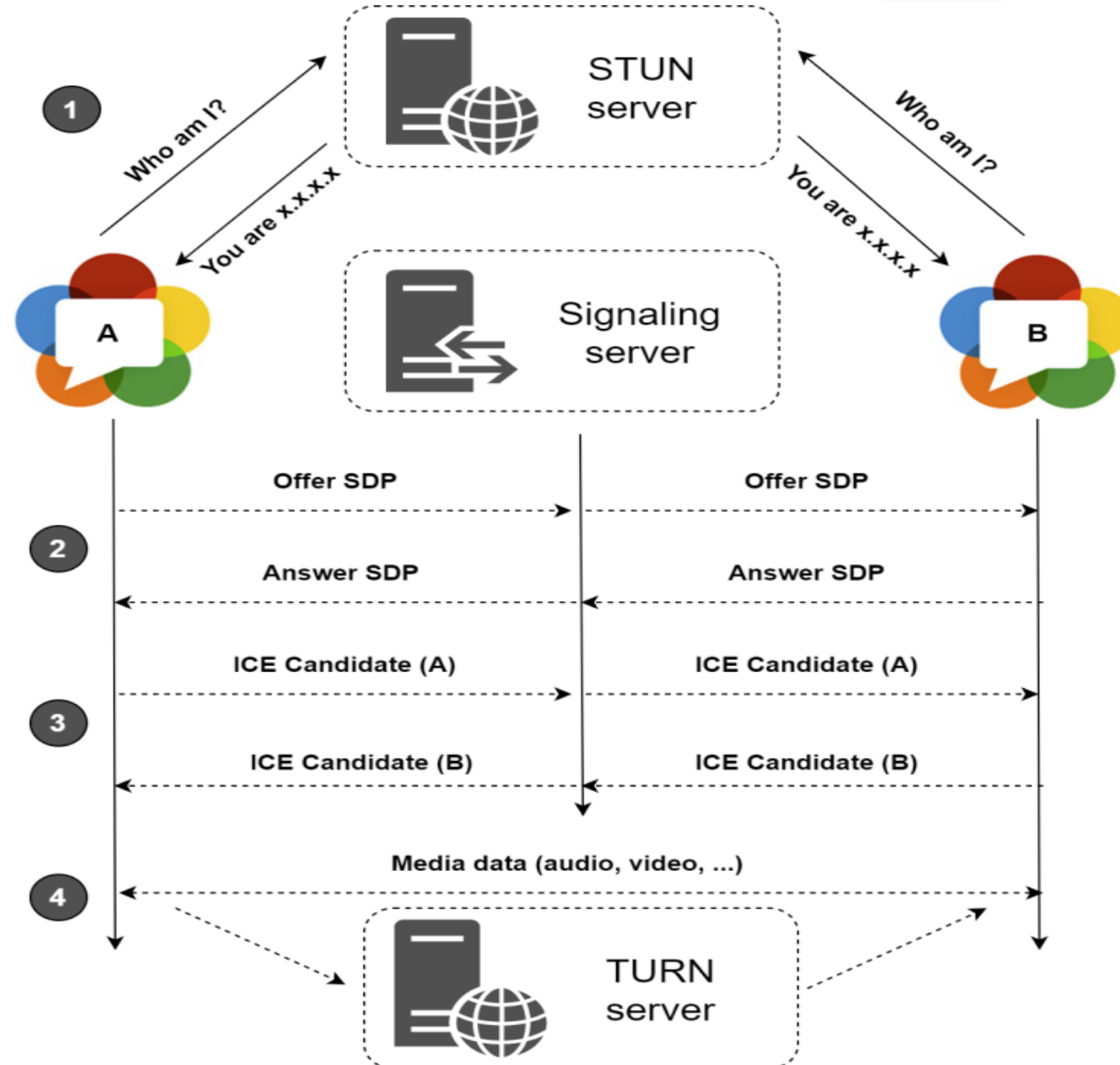


TURN server


## WebRTC connection representation: Scenario with STUN and TURN Servers



## Connection setup Diagram: Step-by-step WebRTC Flow



# Centralisation of Media streams

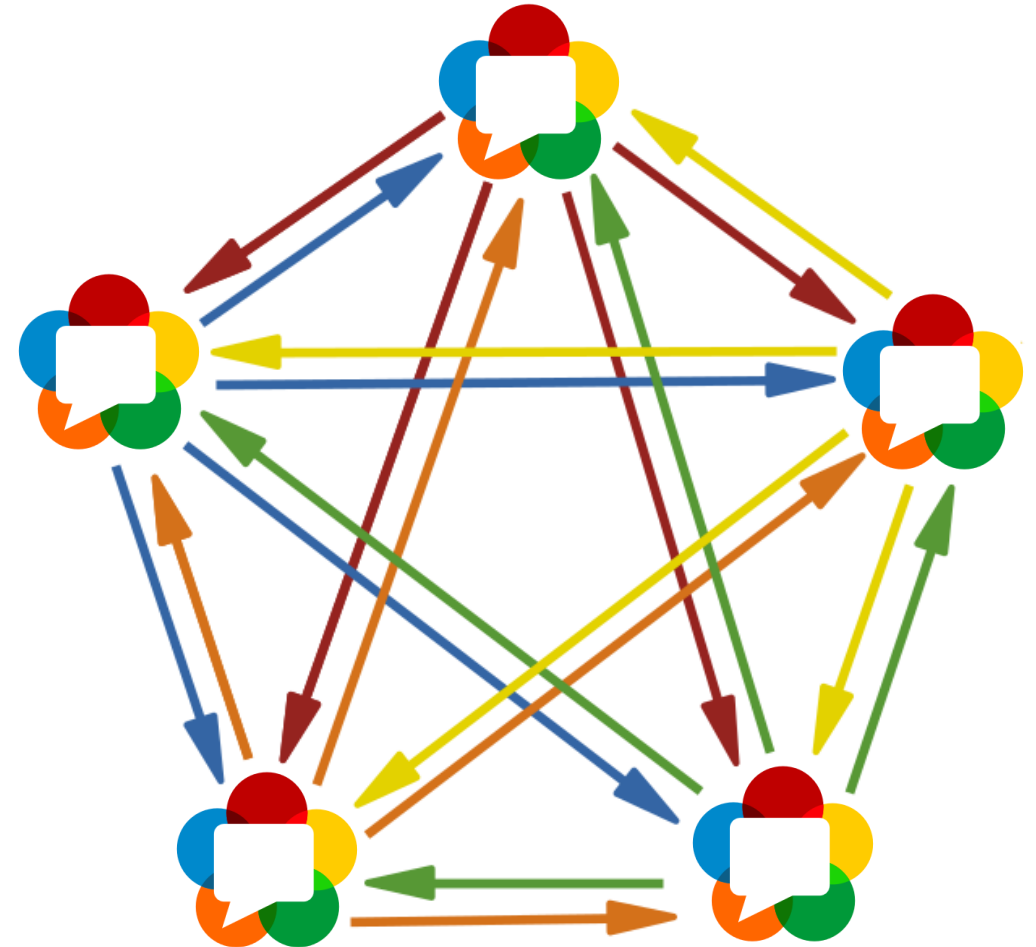
 The implementation of E-Learning  
video conferencing system can rely on  
three main methods of media  
centralization and distribution



## Centralisation of Media streams

## Mesh Topology

This model assumes a decentralized system where streams are transmitted directly between participants. It is used for conferences with a small number of users (approximately 3-5)



## Centralisation of Media streams

### SFU – Selective Forwarding Unit

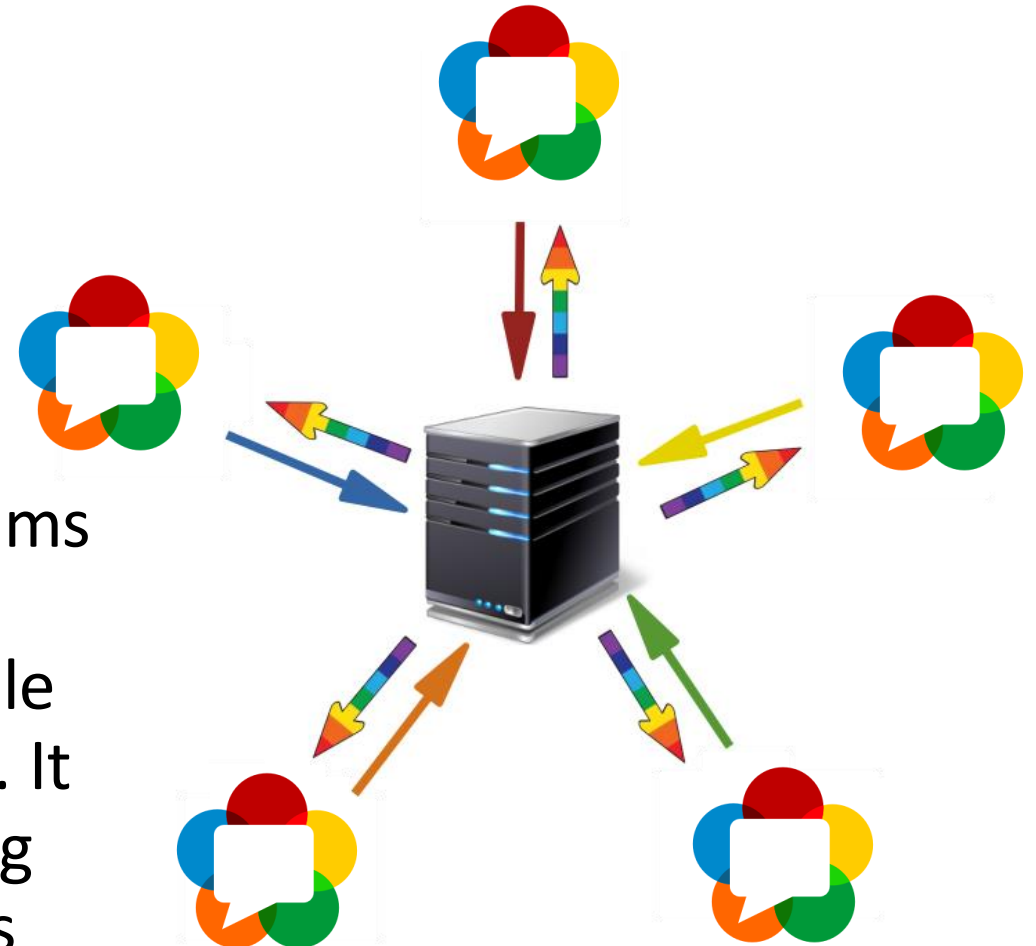
This model centralizes the streams, but not in the most efficient way. It allows connecting multiple participants, but yet it is not well suited for E-Learning platform



## Centralisation of Media streams

### MCU – Multipoint Central Unit

This model not only centralizes the streams but also optimizes their distribution by compositing multiple streams into a single one and retransmitting it to participants. It is optimal solution for video conferencing with a large number of users, which suits our needs perfectly





**Authors:**



**Student:**  
**Topala Ivan**

**Professor:**  
**Tiberiu Socaciu**

