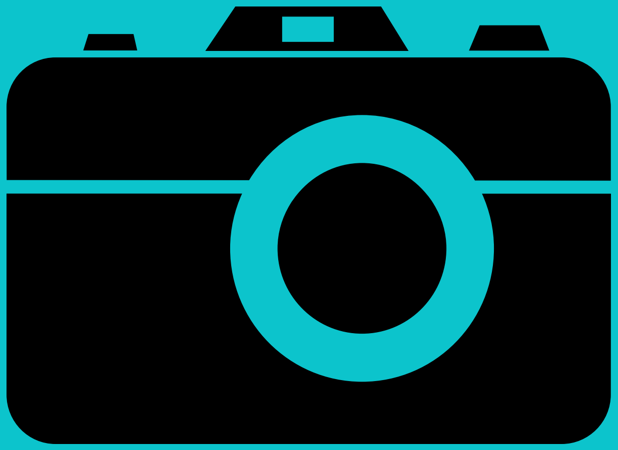
LIVE VIDEO BROADCAST

REAL TIME PLUGIN FREE BROADCAST



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PROJECT REQUIREMENTS

Build a system through which a live video can be broadcasted on internet

- Users can see that broadcast along with the voice in (almost) real time.

- Noise and delay should be minimum.

- Broadcast should be one way.

- Broadcast should be plug-in free.
- (users don’t have to install flash player)

CODEC AND STREAMING TYPE

H.264 -for HD video compression

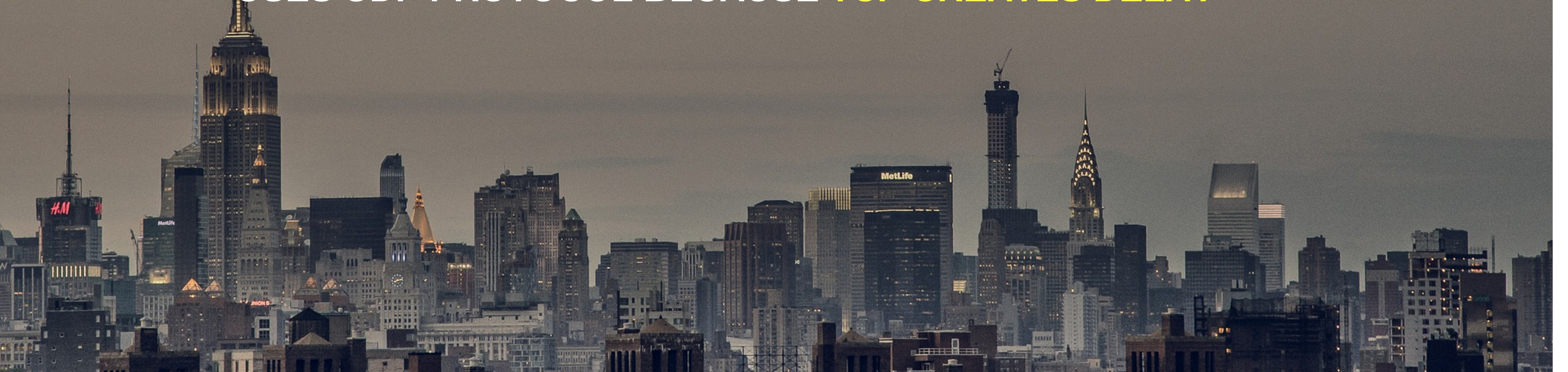
VP8 and VP9 -another video compression technology
-mainly used in WEBRTC applications
-owned by GOOGLE

STREAMING TYPE and PROTOCOL

RTMP / HTTP - TCP-based protocol originally built for Flash
-Streamed video quality is not adjustable with the channel's bandwidth

HLS - Apple-backed protocol that is widely supported on many environments.

WebRTC - Low latency protocol, works in almost all browsers.
-However it is not supported on several environments/browsers Viz. iOS, IE
-USES UDP PROTOCOL BECAUSE TCP CREATES DELAY



EXISTING SOLUTIONS

	SKYPE	HANGOUT	LIVE	USTREAM IBM'S CLOUD VIDEO
DATA USAGE	400KB/S	30-80KB/S	95-110KB/S	400-650KB/S
VIDEO CODEC	VP8 FOR SD VIDEO H.264 FOR HD VIDEO	VP8	H.264 FOR VIDEO ACC FOR AUDIO	H.264
MINIMUM RESOLUTION & FRAMES/SEC	480*270 15-30 FPS	320*240 15FPS	480*270 30FPS	720: 1280X720 1500 KBPS 480: 854X480 1000 KBPS 360: 640X360 750 KPBS 240: 426X240 500 KPBS

Analysis report

WHAT IS WEBRTC

WEB +RTC (REAL TIME COMMUNICATION)

WebRTC is a technology and newly proposed open standard that adds real time communication to web browser.
(WITHOUT THE NEED TO INSTALL PLUGIN)

WebRTC is an open source project supported by Google mozilla and opera.

WEBRTC TECHNOLOGY IS USED BY GOOGLE HANGOUT AND FACEBOOK LIVE

HOW DOES WEBRTC REALLY WORK ?

AN OVERVIEW

FOR MORE INFORMATION PLEASE VISIT
HTTP://WWW.INNOARCHITECH.COM/WHAT-IS-WEBRTC-AND-HOW-DOES-IT-WORK/

QUICK OVERVIEW OF THE KEYWORDS

- Signaling, Sessions, and Protocols

network discovery and NAT traversal.

session negotiation and establishment:

Session Initiation Protocol(SIP)
SDP Protocol

ICE protocol:

IP address, port, and transport protocol to be used

Peer to peer communication:

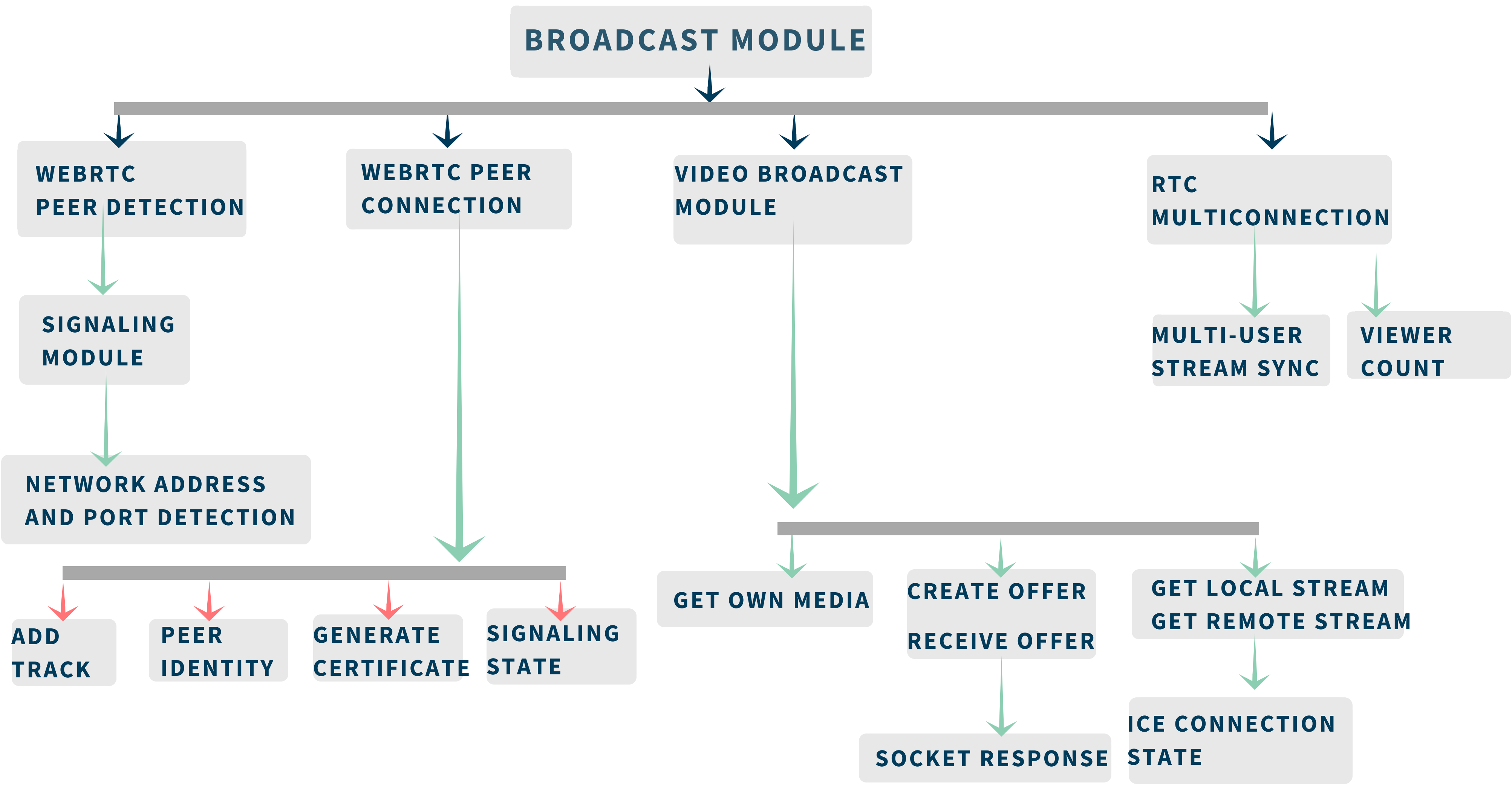
network socket connection

firewall and network access translation device (NAT)

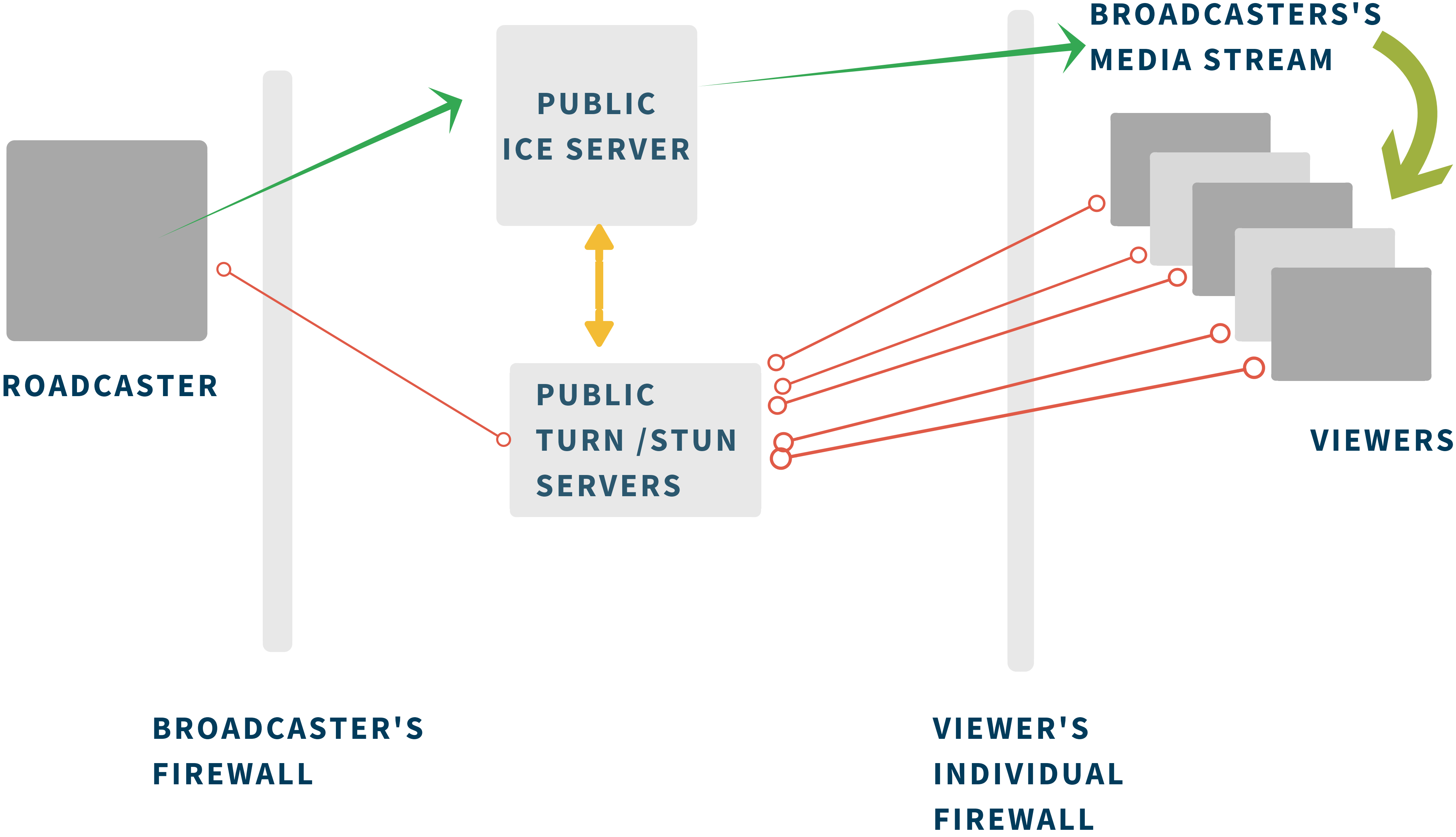
From a very high level, a NAT device translates private IP addresses from inside a firewall to public-facing IP addresses.

TURN(Traversal Using Relays around NAT) AND
STUN(Session Traversal Utilities for NAT) SERVER

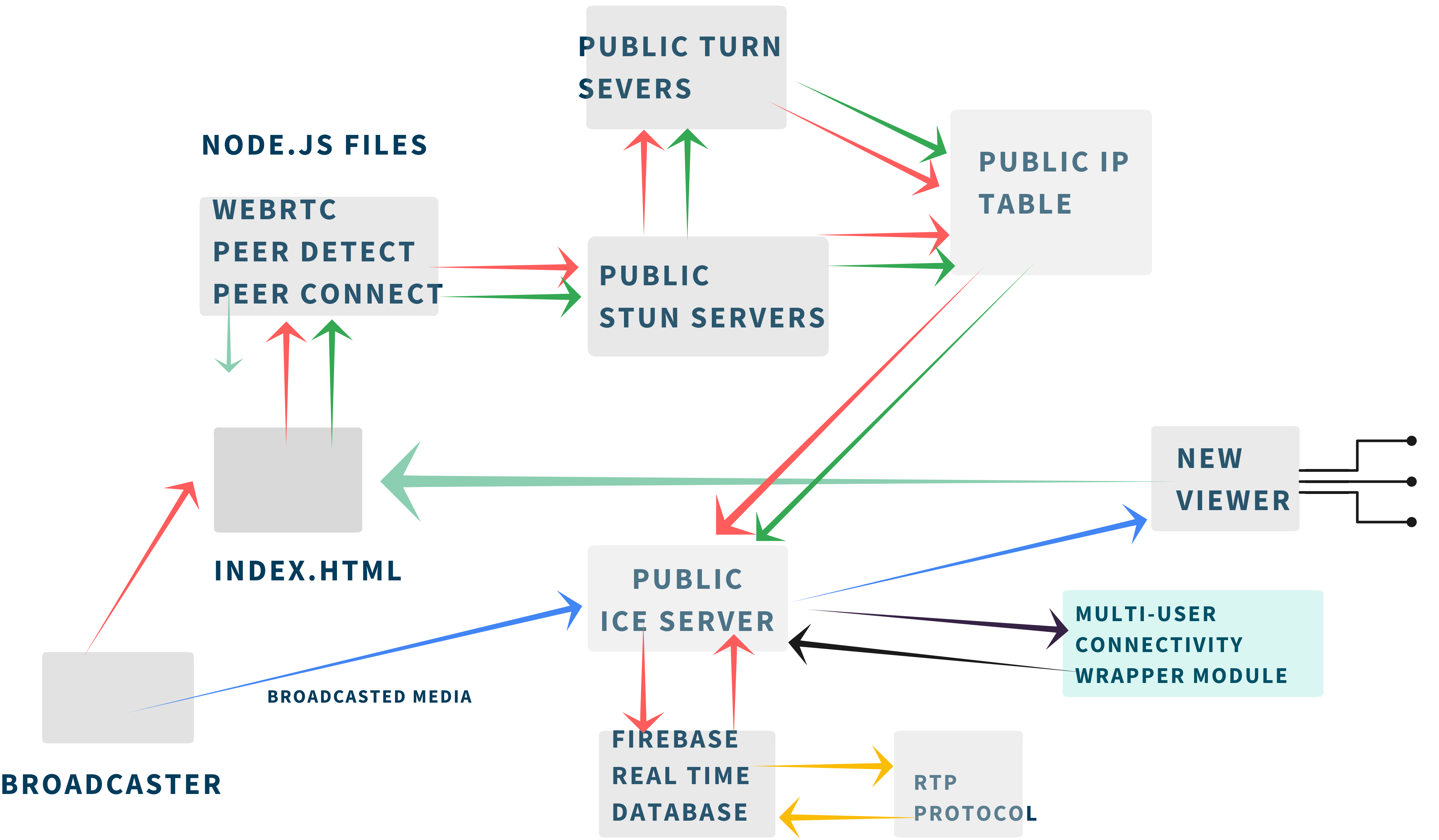
LOGICAL VIEW OF OUR PROJECT



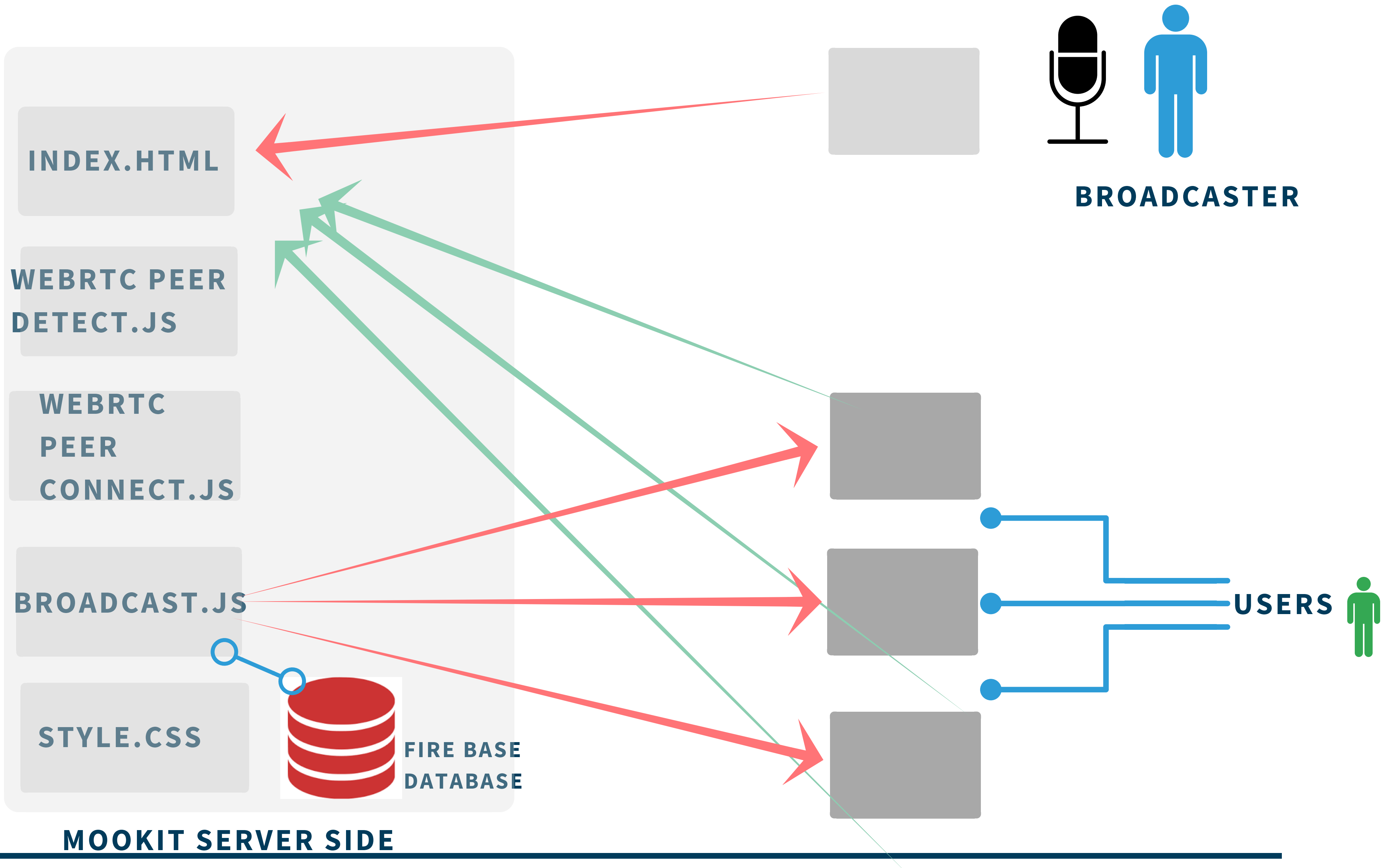
BASIC OVERVIEW OF CONNECTIVITY



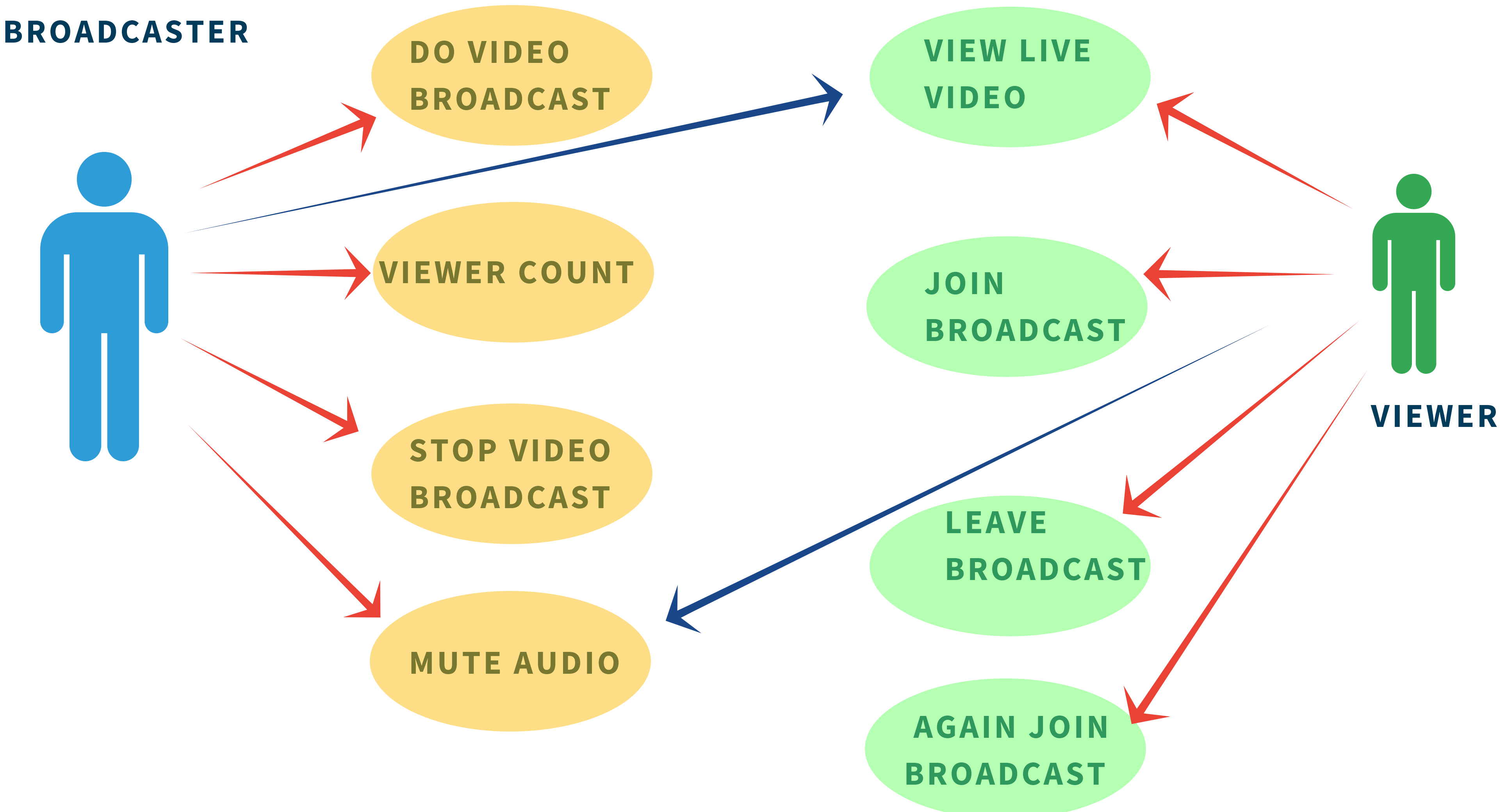
PROCESS VIEW OF OUR PROJECT



DEPLOYMENT VIEW OF OUR PROJECT



USERVIEW OF OUR PROJECT



T E C H N O L O G I E S T E S T S & R E S U L T

- NODE.JS, HTML ,FIREBASE (REAL TIME DATABASE) ,WEBRTC

TESTS & RESULTS

- We have deployed our module on MOOKIT SERVER and tested(on Mac desktop in media labs) for approx 100 users(on 10 machines with 10-10 browser window each) .
- Each user takes approx 30-40 kbps of bandwidth.

- We have broadcasted using 4g mobile
- the broadcasted video is significantly delay-less and continuous(no hang of frames)
- Module is working in approx real time (in worst case scenario 1-2 sec delay is there*)
- (delay depends upon the network of broadcaster as well as receiver)
- Problem of echo is resolved (unless receiver and broadcaster are not at the same place)

LIMITATIONS AND FUTURE WORK

- ON IOS BASED DEVICES WE ARE FACING PROBLEMS WITH CONNECTIVITY
- CHROME IS THE MOST COMPATIBLE BROWSER WE FOUND IN OUR TESTS
- ONLY HTML5 BASED BROWSER ARE ELIGIBLE FOR WEBRTC SUPPORT
- EFFICIENT AND PAID MEDIA SERVER WILL ENHANCE THE CAPACITY OF THE HANDLING THE NUMBER OF VIEWERS

- Thank you !!