WEBRTC VIDEO CHAT ON FIREBASE

How WebRTC Works

Initial Setup

Video Chat Feature

Peer Connection

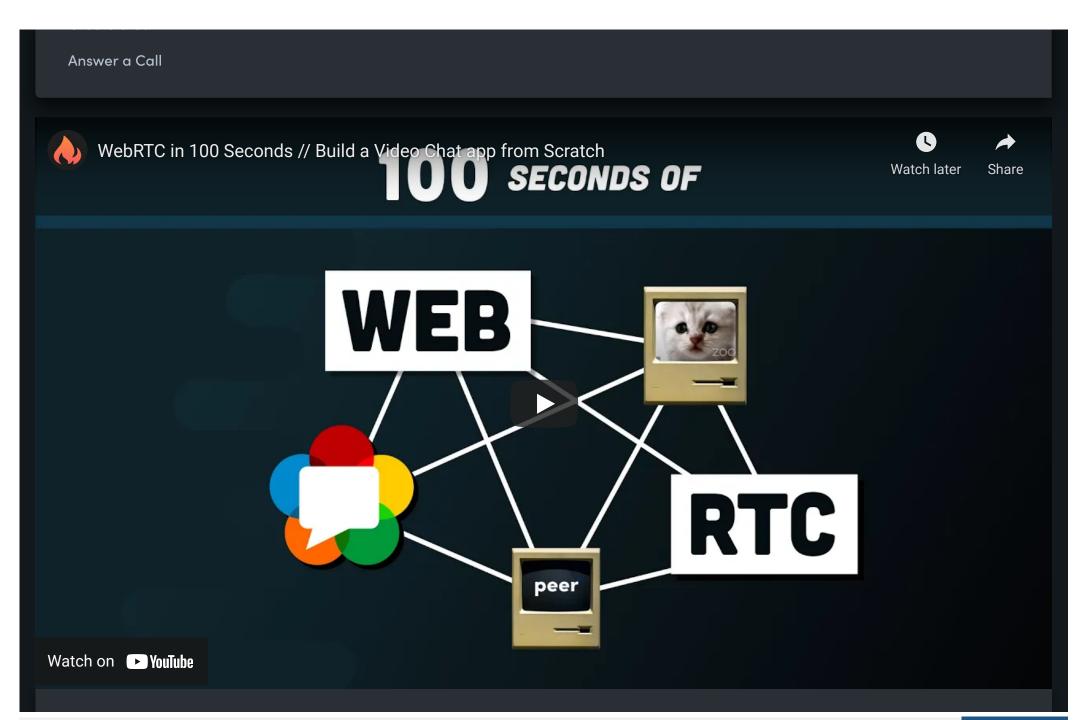
Local Video Stream

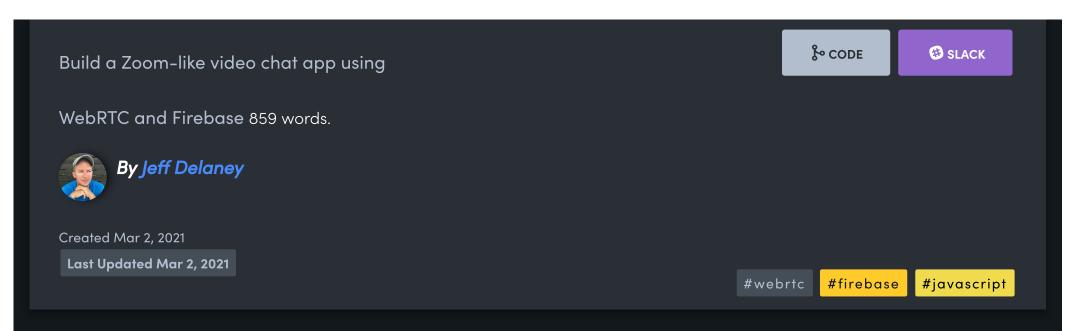
Remote Video Stream

Signaling

Data Model

Create a Call





WebRTC facilities realtime audio/video communication on the web using a peer-to-peer protocol, allowing you to build apps like Zoom, Skype, etc.

The following lesson builds a 1-to-1 video chat, where each peer streams directly to the other peer – there is no need for a middle-man server to handle video content. However, a 3rd party server is required for **signaling** that stores shared data for stream negotiation. Firestore is an excellent choice for WebRTC because it is easy to listen to updates to the database in realtime.

Additional Resources:

Firebase WebRTC Codelab

• Demo with Firebase RealtimeDB

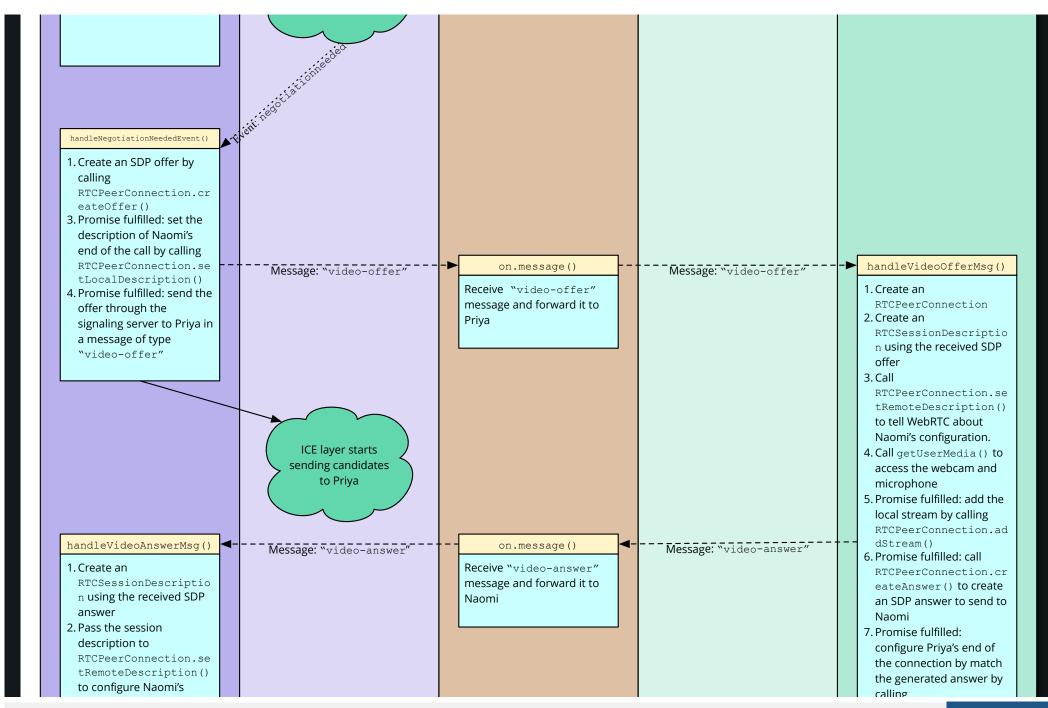
HOW WEBRTC WORKS

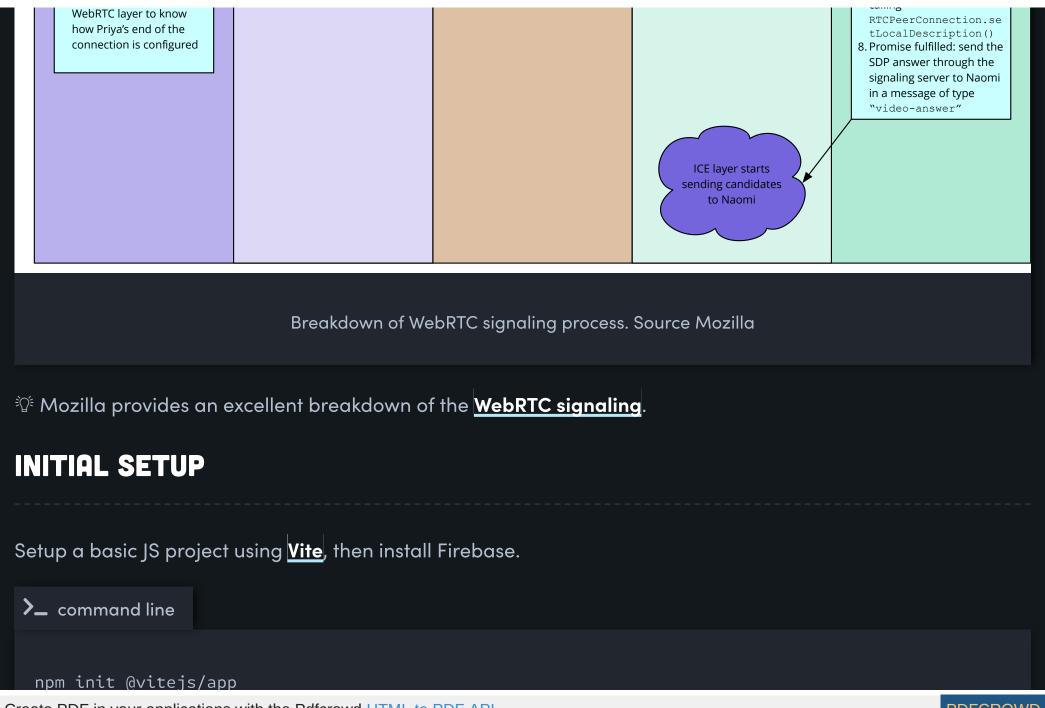
It helps to first understand the steps involved in a WebRTC video chat app from the perspective of the caller and callee.

- 1. Start a webcam feed
- 2. Create an 'RTCPeerConnection' connection
- 3. Call createOffer() and write the offer to the database
- 4. Listen to the database for an answer
- 5. Share ICE candidates with other peer
- 6. Show remote video feed

- 1. Start a webcam feed
- 2. Create an 'RTCPeerConnection' connection
- 3. Fetch database document with the offer.
- 4. Call createAnswer(), then write answer to database.
- 5. Share ICE candidates with other peer
- 6. Show remote video feed

Naomi (Caller)		Signaling Server	Priya (Callee)	
Web App	Web Browser		Web Browser	Web App
invite() 1. Create an RTCPeerConnection 2. Call getUserMedia() to access the webcam and microphone 3. Promise fulfilled: add the local stream by calling RTCPeerConnection.ad dStream()	Ready to negotiate, so ask the caller to start doing so			





```
npm install firebase
Initialize Firebase with Firestore:
 Js main.js
  import firebase from 'firebase/app';
  import 'firebase/firestore';
  const firebaseConfig = {
  };
  if (!firebase.apps.length) {
    firebase.initializeApp(firebaseConfig);
  const firestore = firebase.firestore();
```

VIDEO CHAT FEATURE

The following code snippets break down the most important concepts when building a video chat feature.

Reference the full source code for the complete project.

PEER CONNECTION

Create global variables for the peer connection and video streams. Notice how the peer connection makes a reference to STUN servers hosted by Google - a STUN server is used to discover a suitable IP/port candidates for establishing a P2P connection.

```
Js main.js
const servers = {
  iceServers: [
       urls: ['stun:stun1.l.google.com:19302', 'stun:stun2.l.google.com:19302'],
    },
  ],
  iceCandidatePoolSize: 10,
};
const pc = new RTCPeerConnection(servers);
let localStream = null;
 let remoteStream = null;
```

LOCAL VIDEO STREAM

Create a video feed from a webcam using the MediaStream interface. Add the stream tracks to the peer connection.

```
webcamButton.onclick = async () => {
    localStream = await navigator.mediaDevices.getUserMedia({ video: true, audio: true });

    // Push tracks from local stream to peer connection
    localStream.getTracks().forEach((track) => {
        pc.addTrack(track, localStream);
    });

    // Show stream in HTML video
    webcamVideo.srcObject = localStream;
}
```

REMOTE VIDEO STREAM

Initialize a remote video feed with an empty stream. Eventually, the stream will be populated when tracks are added to the peer connection.

```
remoteStream = new MediaStream();

// Pull tracks from remote stream, add to video stream
pc.ontrack = event => {
    event.streams[0].getTracks().forEach(track => {
        remoteStream.addTrack(track);
    });
};

remoteVideo.srcObject = remoteStream;
```

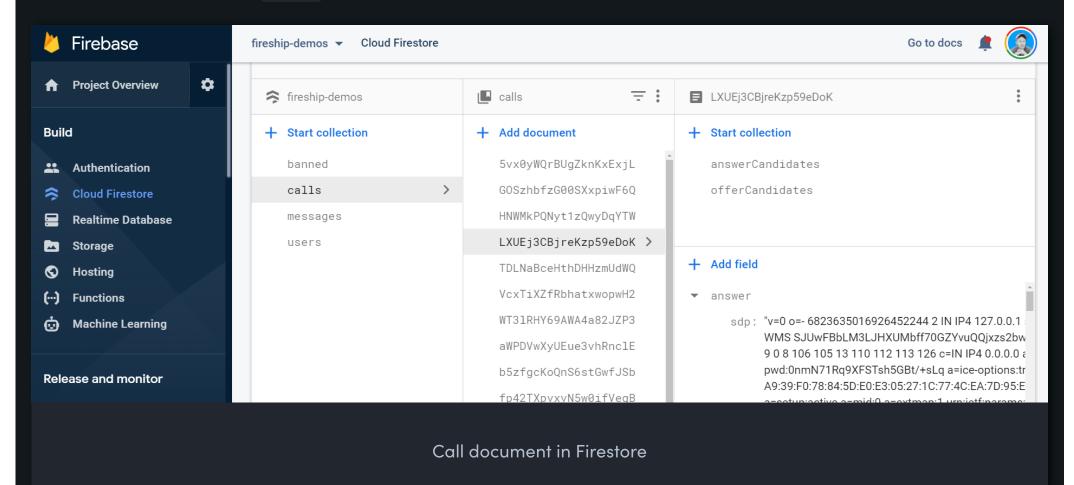
SIGNALING

To connect two or more peers via WebRTC, each clientside app needs to provide an ICE (Internet Connectivity Establishment) server configuration. The apps must **signal** to each other how they should connect, which requires a backend server or database (like Firebase). The database allows the peers to relay the required

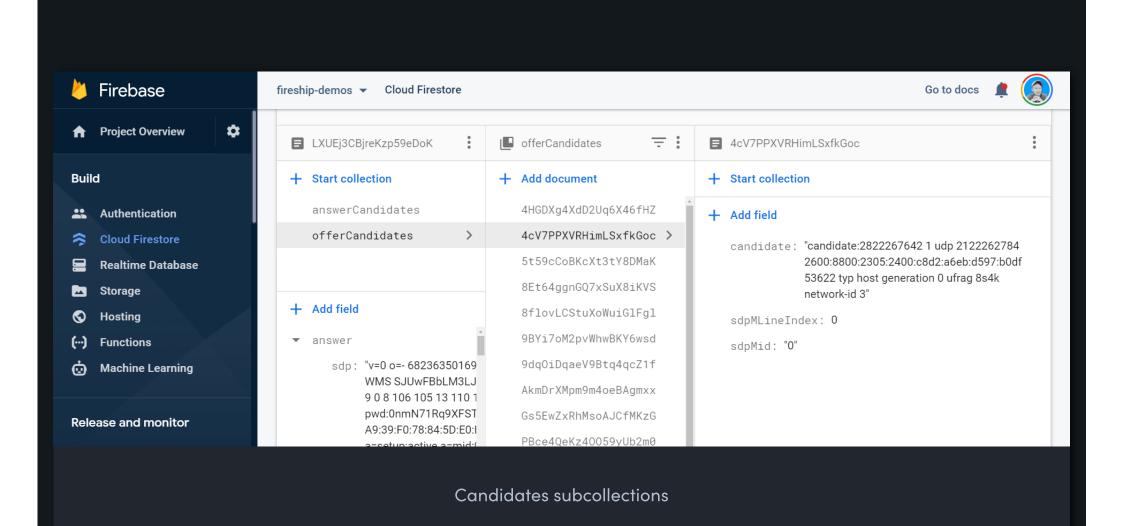
information to establish a connection.

DATA MODEL

The database contains a calls collection that stores the offer/answer objects from each peer.



Each call document contains a subcollection for answerCandidates and offerCandidates.



CREATE A CALL

The caller will reference a new Firestore document with a random ID. The WebRTC peer connection can then

create an offer and write the result to the database

Once created, the caller will then wait for the document to be updated with an answer from the other user.

```
callButton.onclick = async () => {
  const callDoc = firestore.collection('calls').doc();
  const offerCandidates = callDoc.collection('offerCandidates');
  const answerCandidates = callDoc.collection('answerCandidates');
  callInput.value = callDoc.id;
  // Get candidates for caller, save to db
  pc.onicecandidate = event => {
    event.candidate && offerCandidates.add(event.candidate.toJSON());
 };
  const offerDescription = await pc.createOffer();
  await pc.setLocalDescription(offerDescription);
  const offer = {
    sdp: offerDescription.sdp,
    type: offerDescription.type,
```

```
};
await callDoc.set({ offer });
// Listen for remote answer
callDoc.onSnapshot((snapshot) => {
  const data = snapshot.data();
  if (!pc.currentRemoteDescription && data?.answer) {
    const answerDescription = new RTCSessionDescription(data.answer);
    pc.setRemoteDescription(answerDescription);
});
// Listen for remote ICE candidates
answerCandidates.onSnapshot(snapshot => {
  snapshot.docChanges().forEach((change) => {
    if (change.type === 'added') {
      const candidate = new RTCIceCandidate(change.doc.data());
      pc.addIceCandidate(candidate);
  });
});
```

ANSWER A CALL

The process to answer a call is similar, but will reference the existing Firestore document ID, instead of creating a new document.

```
answerButton.onclick = async () => {
  const callId = callInput.value;
  const callDoc = firestore.collection('calls').doc(callId);
  const offerCandidates = callDoc.collection('offerCandidates');
  const answerCandidates = callDoc.collection('answerCandidates');
  pc.onicecandidate = event => {
    event.candidate && answerCandidates.add(event.candidate.toJSON());
  };
  const callData = (await callDoc.get()).data();
  const offerDescription = callData.offer;
  await pc.setRemoteDescription(new RTCSessionDescription(offerDescription));
```

```
const answerDescription = await pc.createAnswer();
  await pc.setLocalDescription(answerDescription);
  const answer = {
    type: answerDescription.type,
    sdp: answerDescription.sdp,
 };
  await callDoc.update({ answer });
  offerCandidates.onSnapshot((snapshot) => {
    snapshot.docChanges().forEach((change) => {
      console.log(change)
      if (change.type === 'added') {
        let data = change.doc.data();
        pc.addIceCandidate(new RTCIceCandidate(data));
   });
 });
};
```



