BlogGeek.me WebRTC Developers Training Courses

* A separate course for support teams is available

January 2020

Tsahi Levent-Levi
tsahil@bloggeek.me

TRAINING OVERVIEW

WebRTC is the most interesting VoIP technology in the last decade. It enables real time voice and video communications from browsers. Companies big and small are using WebRTC to fit into their business processes. The BlogGeek.me WebRTC Training courses are meant to assist in the process of putting the best possible WebRTC architecture in place and successfully developing, launching and maintaining WebRTC applications.

The goal of these WebRTC courses is to bring students up to speed with WebRTC and enable them to make better product decisions, in effect, making them **professional WebRTC developers**.

WHY THESE WEBRTC COURSES?

There's a lot of information about WebRTC on the internet and a few courses and books as well.

The BlogGeek.me WebRTC Training courses are **the only online resources on WebRTC** that are:

1. Up to date with the standard specification and the industry around it
2. Offer a simple, linear view of WebRTC, designed for easier learning experience
3. Include a thorough review of all adjacent technologies
4. Contain a mapping of open source and commercial WebRTC tools

WHO WOULD BENEFIT FROM THESE COURSES?

Anyone who needs to deal with WebRTC as part of his job:

- Developers and QA engineers
- System architects and Lead engineers
- Technical pre-sales, IT, DevOps and Support engineers
- Product and project managers

WHAT'S IN THE BLOGGEEK.ME WEBRTC COURSES?

The courses span 10 training modules with over 60 recorded lessons in total.

One you enroll, you'll be able to work through the modules and lessons by order or skip to the ones that you need at any given time.

- A detailed curriculum can be found later in this document.

- Each course module contains multiple lessons.

- Lessons include a recorded video, textual explanation and links, glossary, lesson brief and Q&A

- Total courses length is about 20 hours of video recordings.

- The courses have last been updated on July 2019.

OFFICE HOURS

Twice a year, office hours are conducted. For a period of ~3 months, 10-12 live office hours will be conducted over WebRTC. The office hours will include additional learning materials to assist students. Office hours are a great opportunity to ask any questions related to WebRTC. They are conducted twice a day on a weekly basis, covering all time zones.

Besides the office hours, students can and are encouraged to use the chat widget to ask questions that arise throughout their training.

ARE THERE ANY PREREQUISITES?

Yes and no.
It is assumed that you have some technical background and inclination around software.
The course has been designed to be suitable for both people who come from the VoIP industry as well as those who have a web background.

ABOUT THE INSTRUCTOR

Tsahi Levent-Levi is an Independent Analyst and Consultant for WebRTC.

Tsahi Levent-Levi has over 20 years of experience in the telecommunications, VoIP and 3G industry as an engineer, manager, marketer and CTO. Tsahi is an entrepreneur, independent analyst and consultant, assisting companies to form a bridge between technologies and business strategy in the domain of telecommunications.

Tsahi is the author and editor of [BlogGeek.me](), which focuses on the ecosystem and business opportunities around WebRTC. Tsahi is the co-founder and CEO of testRTC, a company assisting developers to test and monitor their WebRTC based products. He is also a co-founder of Kranky Geek, a conference for WebRTC for developers, sponsored by Google.

Tsahi serves as a W3C Evangelist for everything WebRTC.

COURSE PLANS

INDIVIDUAL

WebRTC Basics	Advanced WebRTC	WebRTC Tooling	ALL INCLUDED
Free	$600	$300	$800
Modules Introduction	**Modules** Introduction Networking Signaling Codecs Media processing 3rd party Design patterns Bonus lessons	**Modules** Vendor interviews Snippets	**Modules** Introduction Networking Signaling Codecs Media processing 3rd party Design patterns Bonus lessons + Vendor interviews Snippets
Online chat	Online chat	Online chat	Online chat
	Office hours		Office hours

CORPORATE

- Designed for enterprises who need to send multiple employees to the course
- Show corporate logo on course partners page (https://webrtccourse.com/partners/)
- Built around the ALL INCLUDED plan for each participant with additional features:
  - Unlimited yearly access to all materials
  - Show corporate badge of use on website
  - Office hours when a course is active (twice a year)
  - 2 private consultation calls for personal live training

PLAN	SINGLE	CORP 5	CORP 20	CORP 50	UNLIMITED
Plan price	$800	$2,500	$8,000	$15,000	$26,000
Minimum # of students	1	5	20	50	-
Per student price	$800	$500	$400	$300	-

COMPANIES SENDING STUDENTS TO THE ADVANCED WEBRTC ARCHITECTURE COURSE

Join over
500 students
from more than
90 companies

WHAT ARE STUDENTS SAYING?

I've been working with WebRTC for a year before enrolling to this course. After going through the course, I can say that I finally now know exactly how the ICE process works and have a much better understanding of audio and video codecs, media processing and topics such as Simulcast and SVC. I will definitely be getting back to some of the course lessons whenever needed."

Walmyr Filho | QA Engineer | appear.in

The Advanced WebRTC Architecture course is a must-have for anyone interested in WebRTC. Tsahi takes you through everything you need to know about WebRTC at the business and technical architecture level. It's required viewing for all new hires in our team!"

Arin Sime | CEO & Founder | WebRTC.ventures

Advanced WebRTC Architecture course prepared by Tsahi is one of the comprehensible, well-structured and simplified WebRTC sources that we have ever encountered. The course is an invaluable asset for anyone who would like to understand and invest time in WebRTC."

Onur Özen | Director and Head of Product | Adeya

The course is well laid-out and current. I work with this technology in an Ops capacity, so it's great to learn this material more thoroughly rather than piecemeal from my co-workers as I have been."

Heather Young | Dev/Ops & Process Guru | &yet

Our organization is new to WebRTC so many of these concepts were foreign to us. BlogGeek's WebRTC course breaks these concepts up into relatable and easily understood modules which when put all together creates a very clear picture into how WebRTC works. This training has been invaluable to our organization and has equipped our team with enough knowledge and understanding to troubleshoot issues and better support our conferencing platform."

Ryan Cartmell | Director, Production System Administration | Intermedia

This course is very convenient to get a "linear walk-through" of WebRTC, so I don't get lost in a sea of more or less important information. Complementing that with some links for each lesson is also very good."

Fredrik Johansson | Software engineer | Tobii AB

We've signed up our team to the course and the immediate feedback was very positive.
The content and the flow make it easier to learn WebRTC, and there's a good balance for beginners as well as intermediate level of engineers."

Panjak Gupta | Founder and CEO | vCloudX

CURRICULUM (ADVANCED WEBRTC)

MODULE 1 THE BASICS OF WEBRTC

WebRTC Basics
Advanced WebRTC
ALL INCLUDED

A bird's eye view to what WebRTCis and how it works.

This module covers the history of WebRTC as well as its current state. Various touch points of WebRTC and what APIs are available to developers and where will be discussed.

Lessons:

- Introduction – introduction to the course. An explanation to all course modules and the logic behind them
- What is WebRTC? – a look at what is WebRTC and what is it used for
- Browsers and device coverage - understanding where WebRTC is available, on which browsers and mobile devices. What do developers do when they want to use it where it isn't available
- WebRTC APIs - an overview of the JS APIs provided by WebRTC and the basic structure of a peer connection and its media elements

MODULE 2 NETWORKING BASICS

Advanced WebRTC
ALL INCLUDED

WebRTC relies on computer networking. This module explains the various transport protocols with a focus on browser networking and WebRTC related aspects of it, such as NAT traversal. You'll understand why and how ICE is used in WebRTC sessions.

Lessons:

- TCP and UDP - the underlying building blocks of the modern internet. A look at what these transport protocols are and what they are used for
- HTTP - HTTP is how we surf the internet today. This lesson will review how HTTP is used, along with the progress from HTTP 1.0 to today's HTTP/2
- WebSockets - the modern generation of interactive communications in browsers. This transport protocol plays an important role in most WebRTC applications
- NAT Traversal - in this lesson, we cover firewall and NAT limitations on communications and the solutions available to WebRTC: ICE, STUN and TURN
- The 3 layers of VoIP - a philosophical look at today's VoIP applications and where innovation takes place

CURRICULUM (ADVANCED WEBRTC)

MODULE 3

WEBRTC SIGNALING

Advanced WebRTC

ALL INCLUDED

WebRTC has no signaling, but it is still needed for WebRTC to work. Here we will touch the various transport and signaling protocols available to you and discuss topics such as SDP and security in WebRTC. By the end of this module, you will be able to select the signaling module for your application.

Lessons:

- Transport protocols in WebRTC - the underlying communication mechanism used by WebRTC applications is the transport protocol. The various alternatives are reviewed in this lesson
- Signaling protocols in WebRTC - WebRTC offers a wide variety of signaling protocol alternatives. From SIP and XMPP to MQTT as well as proprietary solutions. This lesson will review the options available and which signaling protocol to choose for what use case
- SDP - Session Description Protocol along with JSEP play an important role in connecting media sessions on WebRTC. This lesson covers this topic
- Security - security in WebRTC happens in all layers of the stack. This lesson deals with own security is handled by WebRTC and the application using it
- Screen sharing - how screen sharing is implemented in WebRTC applications

MODULE 4

CODECS

Advanced WebRTC

ALL INCLUDED

To send audio and video means dealing with Codecs. This module covers all you need to know (and more) about the codecs that WebRTC uses. You will be able to pick the codecs you'll want to use for your WebRTC service once we're done with this module.

Lessons:

- Basics of voice codecs - an overview of voice codecs with a special focus on VoIP characteristics of voice codecs
- Opus - explanation of the most important voice codec in WebRTC
- Other voice codecs (and WebRTC) - what other voice codecs are widely used and how to connect them to WebRTC when needed
- Basics of video codecs - an overview of video codecs with a special focus on VoIP and real time communications characteristics of video codecs
- VP8 and H.264 - the mandatory to implement video codecs in WebRTC. Where are they available, and when to use each of these alternatives
- VP9 and the Alliance of Open Media - a look at the future of WebRTC video coding. Here we cover VP9, AV1 and a bit of the politics around codec selection

CURRICULUM (ADVANCED WEBRTC)

MODULE 5 — **MEDIA PROCESSING**

Advanced WebRTC

ALL INCLUDED

Group calling? Recording? These are tough nuggets in media processing. This is why we have this module, which goes through the various multiparty architectures, explains them in detail along with the best approaches of using them.

Lessons:

- RTP and RTCP - how RTP, RTCP and SRTP are used in WebRTC as the media transport protocol. Includes an explanation of multiplexing and bundling of sessions
- Mesh - mesh architectures in group calls
- Mixing (MCU) - Multipoint Conferencing Unit and mixing media inputs in large group calls
- Routing (SFU) - Selective Forwarding Unit and routing media inputs in large group calls. Includes explanation of simulcast and SVC
- Recording - recording alternatives in WebRTC. Includes both server-side and client-side recording solutions. When to choose which option
- Transcoding - how transcoding is done with WebRTC, and recent trends in this area

MODULE 6 — **3rd PARTY FRAMEWORKS AND SERVICES**

Advanced WebRTC

ALL INCLUDED

Here's a secret for you: No one develops using WebRTC from scratch these days. Everyone uses a framework of sorts, be it open source or commercial. After going through this module, you'll understand the ecosystem around WebRTC and be able to pick tools into your WebRTC technology stack.

Lessons:

- Development strategies for WebRTC - how different vendors are tackling development of WebRTC applications
- Signaling alternatives - a review of available open source, commercial and hosted/managed alternatives for WebRTC signaling
- Media server alternatives - an explanation and high-level architecture review of the popular open source media servers for WebRTC
- WebRTC PaaS - CPaaS, Communication Platform as a Service, offering WebRTC capabilities. Who uses them and why
- VoIP frameworks - a look at VoIP specific WebRTC solutions that need to connect to SIP and PSTN
- Testing and monitoring - a look at DIY, open source and commercial alternatives to testing and monitoring, along with challenges and best practices in this domain

CURRICULUM (ADVANCED WEBRTC)

MODULE 7 COMMON WEBRTC DESIGN PATTERNS

Advanced WebRTC
ALL INCLUDED

This module acts as a summary for this course. It holds a lot of common design patterns of applications, so you can go through them and see how all the lessons fit into a single architecture of an application.

Lessons:

- Media flows in WebRTC - examples of different media flows in WebRTC use cases, covering media servers and various NAT traversal scenarios
- Meetings recorder - how to add media recording capabilities to an application. What are the alternatives available? When to use which option
- Multiparty conferences - design of multiparty video service, with a deep dive to UI/UX and screen layout aspects of the service
- Webinar / Low latency live broadcasting - how to build and scale a broadcasting service using WebRTC
- PSTN connectivity - connecting WebRTC to legacy telephony networks

MODULE 8 BONUS LESSONS

Advanced WebRTC
ALL INCLUDED

The "etc" module of this course.

Here, lessons that just didn't fit elsewhere but are highly relevant and important found their place.

They range from requirements, through standardization processes all the way to media algorithms and TURN configuration and deployment.

Lessons:

- WebRTC standardization - explanation of how WebRTC standardization works and what to expect of it moving forward
- Writing RFP requirements for WebRTC - a suggested workflow and template for creating requirements for WebRTC applications
- Media algorithms - a look at adaptive jitter buffers, echo cancellation and other media algorithms used in VoIP and WebRTC
- Mini: WebRTC server-side basics - the free mini course on WebRTC servers
- Mini: H.264 vs VP8? - the free mini course on selecting a video codec for WebRTC applications
- Mini: Effectively connecting WebRTC sessions - how to connect more of your sessions properly
- WebRTC 1.0 - a webinar recording about the changes made to WebRTC 1.0
- Live: Video quality in WebRTC - a webinar recording of aspects related to video quality of WebRTC applications
- Live: Deploy (co)TURN on AWS - a webinar recording of deployment configuration for COTURN servers on AWS

CURRICULUM (WEBRTC TOOLING)

MODULE A	IN 10 MINUTES OR LESS

WebRTC Tooling

ALL INCLUDED

A collection of focused interviews with the people behind the most popular WebRTC tools used by developers:

- Janus - Lorenzo Miniero
- Jitsi - Saúl Ibarra Corretgé
- Kurento - Juan Navarro
- mediasoup - Iñaki Baz Castillo and José Luis Millán
- openVidu - Pablo Fuente
- adapter.js - Philipp Hancke
- Microsoft WebRTC in UWP - James Cadd
- Pion WebRTC - Sean Dubois
- Ant Media - Ahmet Oguz Mermerkaya
- Frozen Mountain - Anton Venema
- Red5 Pro - Dima Nazarenko

MODULE B	SNIPPETS

WebRTC Tooling

ALL INCLUDED

A growing collection of short lessons (3-8 minutes each) covering specific topics in an actionable format.

Snippets:

- WebRTC session disconnections
- ICE configuration
- A quick review of QUIC
- Synchronizing data channel and video
- What's the difference between Plan B and Unified Plan
- How does WebRTC estimate bandwidth?
- HTTP/2 or WebSocket
- Fault tolerance and high availability
- mDNS and <host>.local ICE candidates
- Best practices in media server location
- ICE, Trickle ICE, ICE lite and ICE-TCP
- What and how to log in WebRTC
- How are ICE connectivity checks conducted?
- Resolutions and bitrates for codecs