WebRTC School

Learn and Qualify

WebRTC School Qualified Integrator (WSQI™) program

Overview

The WebRTC School™ is 'the' place to learn all about WebRTC, also known as Web Real-Time-Communications. There is so much information on the internet about WebRTC that is hard to read, poorly presented and also lacking in detail, making it difficult for people to learn about this most important specification. So the WebRTC School™ with its lively, clear and fully animated eLearning program has become the best place to enroll to learn about WebRTC.

Who would benefit from the WebRTC School Qualified Integrator (WSQI™) program?

People who need to understand the underlying infrastructure that helps to make the WebRTC 'magic' happen. Covering protocols, media flows, signaling, NAT traversal techniques and security plus various use case scenarios this program can be taken as a standalone course or used to complement the WebRTC School Qualified Developer (WSQD™) program.

IMPORTANT!!! Pre-requisite knowledge

Before you purchase this course please note that it assumes that you have a 'good' knowledge of Data networking and Voice over IP. If not, then a good place to start is the **Networking for VVoIP** program from The SIP School – www.thesipschool.com

What's in the WebRTC School Qualified Integrator (WSQI™) program?

Once you've enrolled, you'll see 10 modules. You can work through the modules in order or simply choose the ones you are most interested in. The modules are listed here but for more detail, please look further into this document.

1. Introduction to WebRTC
2. Media Flows in WebRTC
3. WebRTC Protocols
4. IETF WebRTC Standards effort
5. WebRTC Media Stack
6. Signaling
7. WebRTC NAT Traversal
8. Security
9. WebRTC 'Use Cases'
10. Status of WebRTC and 'What's next'

Total running time: 3 hours.

Total Running time for this program is approximately 3 hours from the start to finish though the time will vary based on the student's own experience and of course, how much time they want to spend on the material and if they want to replay some modules.

This does not include study time for the WSQI™ test or the taking of the WSQI™ final test itself.

Become a 'WebRTC School Qualified Integrator' or WSQI™

You can gain access to the test *separately* or as part of the main training package – check license 'purchase' options carefully.

The **WSQI™** certification is recognized 'Globally' and is supported by the TIA (Telecoms Industry Association) along with WebRTC World, Comptel, USTelecom and a rapidly growing number of Manufacturers, Service providers and Carriers.

To prepare for the certification test, most training modules have their own 'mini' quiz at the end to help delegates 'gauge' how well they are doing.

NOTE: An access license for any training course and certification test is for 12 months from the date of purchase and if a test is purchased it must be taken before the license period expires.

Introduction to WebRTC

Module times

- Running time = 14 minutes
- Quizzes = 2 minutes
- Total = 16 minutes

Topics Include:

- Real-Time Communication on the Internet
- WebRTC is “Skype™ in the browser”
- What's New?
- A Short History of WebRTC
- WebRTC Support of Multiple Media
- WebRTC Triangle
- WebRTC Trapezoid
- WebRTC and SIP
- WebRTC and Jingle
- WebRTC and PSTN

Media Flows in WebRTC

Module times

- Running time = 9 minutes
- Quizzes = 2 minutes
- Total = 11 minutes

Topics Include:

- Media Flows in WebRTC
- Media without WebRTC
- Peer-to-Peer Media with WebRTC
- NAT Complicates Peer-to-Peer Media
- What is a NAT?
- NAT Example
- NATs and Applications
- Peer-to-Peer Media ‘through’ NAT
- ICE Connectivity Checks
- P2P Media Can Stay Local to NAT
- ICE Servers
- Browser Queries STUN Server
- TURN Server Can Relay Media
- NAT and IPv6

WebRTC Protocols

Module times
- Running time = 18 minutes
- Quizzes = 2 minutes
- Total = 20 minutes

Topics Include:

- What's New?
- WebRTC: A Joint Standards Effort
- IETF Standards
- The WebRTC Protocol Stack
- WebRTC Protocols
- Hypertext Transport Protocol
- The WebSocket Protocol
- Secure Real-Time Transport Protocol
- Session Description Protocol
- Interactive Connectivity Establishment
- Session Traversal Utilities for NAT - STUN
- Transport Layer Security
- Datagram Transport Layer Security
- Stream Control Transport Protocol
- Transmission Control Protocol
- User Datagram Protocol
- Internet Protocol
- What about SIP?

IETF WebRTC Standards effort

Module times
- Running time = 9 minutes
- Quizzes = 2 minutes
- Total = 11 minutes

Topics Include:

- RTCWEB Working Group Documents
- RTCWEB Work in Other WGs
- WebRTC Use Cases and Requirements
- Codec Overview
- Opus IETF Audio Codec
- Opus IETF Audio Codec
- Video Codecs
- Data Channel Usage
- Data Channel Protocols

WebRTC Media Stack

Module times

- Running time = 8 minutes
- Quizzes = 2 minutes
- Total = 10 minutes

Topics Include:

- RTP Header
- RTCP - RTP Control Protocol
- SAVPF Profile of RTP
- Multiplexing RTP and RTCP
- Multiplexing Voice & Video
- Symmetric RTP
- RTP Extension Summary

Signaling

Module times

- Running time = 29 minutes
- Quizzes = 3 minutes
- Total = 32 minutes

Topics Include:

- Role of Signaling
- Why Signaling is Not Standardized
- Server Chooses Signaling Protocol
- Some Signaling is Needed
- Some signaling does need to be standardized
- Signaling in WebRTC
- Signaling State Machine
- Privacy in WebRTC
- Signaling Transport Example: WebSockets
- Signaling Transport Example: HTTP
- Signaling Transport Example: Data Channel
- Signaling Protocol Options
- Signaling Example: HTTP Polling
- Signaling Example: WebSocket Proxy
- Signaling Example: SIP
- SIP over WebSockets
- SIP Usage in WebRTC
- SIP over WebSocket Example
- Open Source JavaScript SIP Stacks
- Jingle over WebSockets
- Signaling Example: Jingle
- Jingle over WebSockets Example
- Open Source JavaScript XMPP
- Signaling over an Overlay Example
- Comparison of Approaches
- Offer Answer Negotiation
- JavaScript Session Establishment Protocol (JSEP)
- Provisional Answers
- JSEP State Machine
- Offer/Answer
- Session Description Protocol (SDP)
- SDP Example
- Rules for Making an Offer
- Rules for Generating Answer
- Simple Audio Video Example

WebRTC NAT Traversal

Module times

- Running time = 39 minutes
- Quizzes = 3 minutes
- Total = 42 minutes

Topics Include:

- Introduction to NAT traversal
- ICE Call Flow
- Gathering Candidates
- Gathering Candidates: STUN
- STUN Operation
- STUN Message Header
- STUN Attribute Format
- STUN Attributes
- STUN Security
- STUN Transport
- Gathering Candidates: TURN
- High Level TURN Call Flow
- TURN Methods
- TURN STUN Attributes
- TURN Security
- TURN Transport
- TURN ChannelData Message
- STUN Error Codes
- TURN Call Flow with Authentication
- Exchange Candidates
- Connectivity Checks
- ICE usage of STUN
- Connectivity Checks: States
- Generation of New Candidates
- ICE Security
- Choose Pair
- Send Keepalives
- ICE Restart
- ICE Lite
- ICE Lite Call Flow
- Trickle ICE

Security

Module times

- Running time = 29 minutes
- Quizzes = 3 minutes
- Total = 32 minutes

Topics Include:

- What are Security and Privacy?
- Browser/Web Security Model
- How WebRTC Changes Browser Privacy
- Browser Prompts for Permission
- How WebRTC Changes Browser Security
- New Attacks
- WebRTC API Attacks
- Signaling Channel Attacks
- Security of WebRTC Media Session
- Website Identity
- Browser User Identity
- SRTP
- Secure Profile of RTP (SRTP)
- SRTP Key Management
- Key Exchange in Signaling
- SDP Security Descriptions
- Generating a Key in Media Path
- SRTP Call Flow
- DTLS-SRTP key agreement
- DTLS Client/Server
- Authenticating a Key Agreement
- Authenticating a Fingerprint
- Identity Proxy in WebRTC Triangle
- Single Identity Proxy with Triangle
- Identity Proxy in WebRTC Trapezoid
- Identity Services
- Communication Consent
- ICE Communication Consent
- Privacy in WebRTC
- Identity Privacy
- IP Address Privacy
- Browser Fingerprinting
- Media Privacy
- WebRTC and the Enterprise

WebRTC ‘Use Cases’

Module times

- Running time = 7 minutes
- Quizzes = 0 minutes
- Total = 7 minutes

Topics Include:

- WebRTC Use Cases and Requirements
- Web Conferencing (Multiparty Conferencing)
- Communications Client (UC and Consumer)
- Contact Centers (B2C and Agent)
- Distributed Communication (Freemium Services)
- Mobile (Voice for Smartphones)
- Single Line of Code WebRTC (Libraries)
- Control (Microphone and Cam Access)
- Gaming (In-Game Media, Chat and Data Channel)
- Overlay Network (Data Channel)

Status of WebRTC and ‘What’s next’

Module times

- Running time = 5 minutes
- Quizzes = 0 minutes
- Total = 5 minutes

Topics Include:

- Status of WebRTC APIs
- Status of WebRTC Protocols
- Status of Browser support of WebRTC
- Support of Signaling Channel
- Support in Mobile
- Opportunities
- Obstacles
- What’s next?