WebRTC School
Learn and Qualify

WSQI™ Certification

Become a **WebRTC School Qualified Integrator (WSQI™)** supported by the Telecommunications Industry Association (TIA)

Exam Objectives

The **WebRTC School Qualified Integrator (WSQI™)** is designed to test your skills and knowledge on the underlying infrastructure that helps to make the WebRTC 'magic' happen. Everything that you need to cover in order to pass this test is covered in the **WebRTC School Qualified Integrator** program but if you decide to learn about WebRTC elsewhere then these are the topics that you should learn about in order to be prepared for the test.

This list is the same as the 'course topics' list also found under the 'outline' button next to the course name in the Catalog.

Please note that if you go along an alternate training path it is possible that you may get a question that may not have been covered in that path. It's up to you!

Please view the following pages for the complete topic list….

Introduction to WebRTC

Topics to study:

- 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

Topics to study:

- 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

Topics to study:

- 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

Topics to study:

- 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

Topics to study:

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

Signaling

Topics to study:

- 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

Topics to study:

- 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

Topics to study:

- 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’

Topics to study:

- 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’

Topics to study:

- 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?