

Technical requirements for the virtual room

WebRTC is used for audio and video transmission.

WebRTC (Web Real Time Communications) is a future-oriented, open standard at W3C that enables real-time communication natively via a web browser without additional software and plug-ins. Low latency and the peer-to-peer connection under https make it very efficient and tap-proof.

You can verify that you meet these requirements with **our test server**:

1. Open the following URL in your web browser. Please use latest versions of Google Chrome, Mozilla Firefox, Microsoft Edge or Safari, if you are on iOS (iPad and iPhone):

<https://virtualroom3.de/index-eco-test.html>

2. Enter your name on the loaded web page and click „Join“.
3. You are asked to authenticate yourself to access the virtual room. Please enter the following information and click „OK“.

Benutzer: test-user-eco

Password: 3duP1W!t

4. After entering the virtual room follow the instructions to activate your microphone. Test your audio and video connection by clicking the corresponding icons you see at the center bottom of your virtual room.

Be advised that there may be other persons also testing this virtual room.

If you are unable to enter the virtual room or are experiencing audio or video problems, please check the following technical requirements.

For a smooth WebRTC communication the following requirements must be met:

1. Please use the current versions the following web browsers
 - Google Chrome or Mozilla Firefox (on MacOS)
 - Google Chrome, Mozilla Firefox or Microsoft Edge as of version 83.0.478.37 (on MS Windows)
 - Safari on iOS (iPad and iPhone)

2. **Check whether the required permissions are set correctly in the web browser**

This is only necessary if the popups for access to webcam etc. do not show up and instead a message is displayed that the access was denied.

Firefox:

- Preferences -> Privacy and security -> Permissions (Camera, Microphone ...) -> Settings
- Check whether the website is in the list and whether access to the devices is permitted
- **Uncheck "Block new requests asking to access your camera"**

Chrome:

- Preferences -> Privacy and security -> Site-Settings -> Permissions
- **"Ask before accessing" must be enabled for each device**

Safari:

- Preferences -> Websites
- Check for each device

MacOS in general:

- Here access to the devices is also strictly controlled by the operating system.
- If a device doesn't work:
 - System preferences -> Security -> Privacy
 - Check whether the browser is basically allowed to use the necessary devices

iOS

- Preferences -> Safari -> Website settings -> Camera / Microphone -> set to "Ask" at least

3. **Ensure that WebRTC is not blocked in the web browser (WebRTC is not blocked by default)**

You can check the setting and change it if necessary:

Firefox:

1. Enter **about:config** in the address bar
2. Click the "I Take the Risk" button that appears
3. Enter media.peerconnection.enabled in the search bar
4. If the value is on **"false"** please double-click to set the value to **"true"**

Chrome:

Several Chrome extensions are known that could block WebRTC, such as uBlock Origin and WebRTC Network Limiter:

1. uBlock Origin:

chrome-extension://cjpalhdlnbpafiamejdnhcphjbkeiagm/dashboard.html#settings.html

Please enable the WebRTC here if necessary - this is the case by default.

2. WebRTC Network Limiter

If the WebRTC Network Limiter is in use, please select the following option from the "Options" menu: **"Give me the best media experience"**

4. **The following ports must be opened**

Ports	Protocol	Description
80	TCP	HTTP
7443	TCP	HTTPS
443	TCP/UDP	TLS listening port (TURN over TLS)
3478	TCP/UDP	Coturn listening port (STUN)

16384 - 32768	UDP	WebRTC, FreeSWITCH, Kurento, HTML5 RTP streams
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- **Access to the following servers for the virtual rooms:**
bilduin-classroom.de, virtualroom1.de to virtualroom12.de
- **Access to TURN/STUN servers:**
yannyana-turn.de, turn-1.de to turn-5.de

5. **VPN client or proxy does not block WebRTC**

Please check whether WebRTC is being blocked by your VPN clients and / or your proxy.

6. **Proxy Server does not interfere with HTML5 WebSocket communication**

HTML5 web socket communication leads to an enormous reduction in unnecessary network traffic and latency compared to conventional polling and long polling solutions.

Proxy servers usually work fine with WebSockets. In some cases, additional proxy server configuration or an update of the proxy server may be necessary for smooth communication.

You can test the WebSocket and WebRTC compatibility of your browser using the following links:

<https://test-webrtc.bilduin.de/>

<https://websocketstest.com/>

<https://html5test.com/>

7. **If necessary disable SSL scanning**