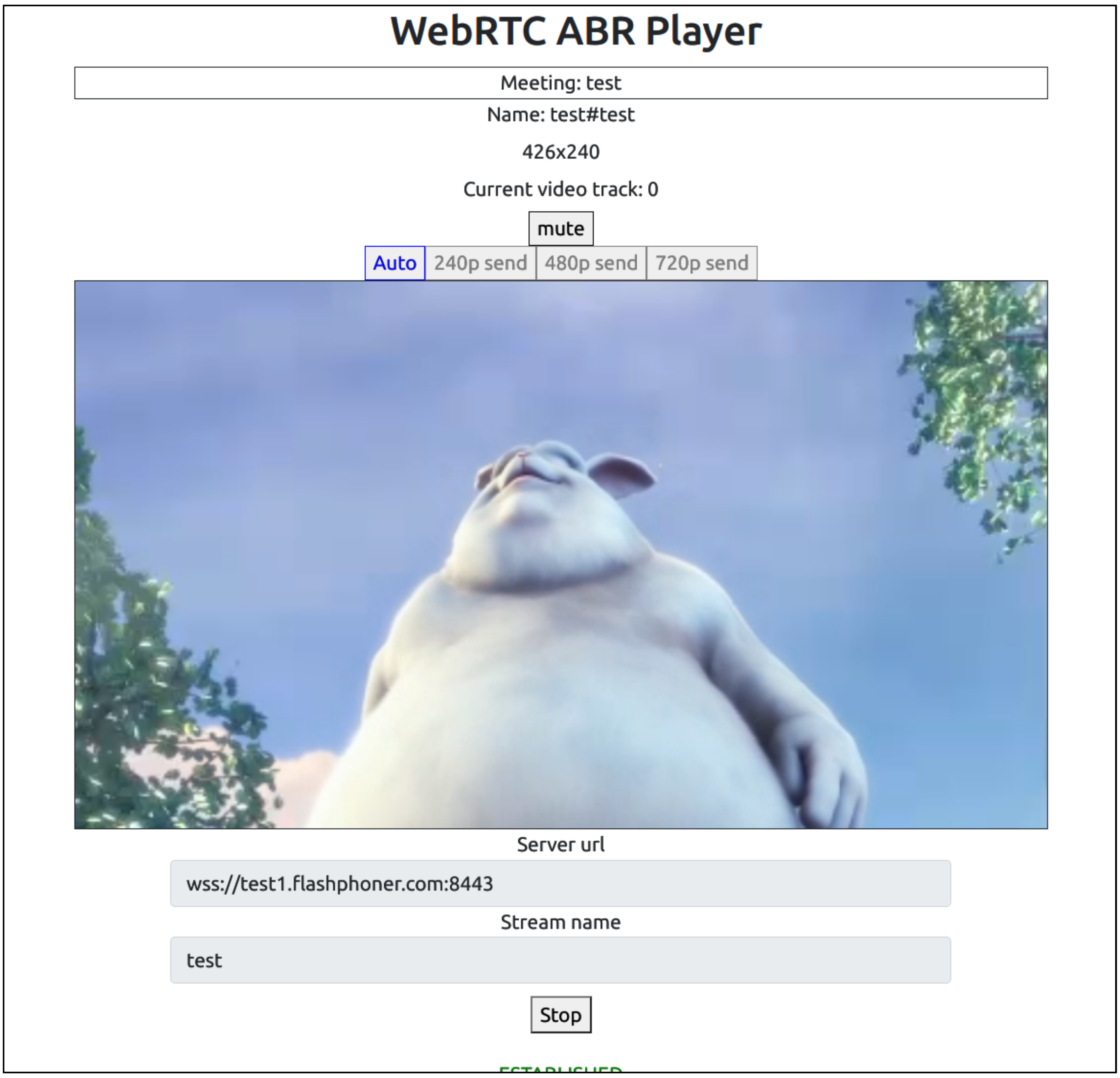
WebRTC ABR Player

The example shows how a stream published to WCS server may be played in a number of video qualities via WebRTC.

On the screenshot below:

- Server url - WCS server websocket URL
- Stream name - stream name to play
- Auto, 240p send, 480p send, 720p send - quality switching buttons named by quality profiles from `/usr/local/FlashphonerWebCallServer/conf/wcs_sfu_bridge_profiles.yml` file



Note that audio track is playing separately

Example source code

The source code consists of the following modules:

- player.html - HTML page
- player.css - HTML page styles
- player.js - main application logic

Analyzing the code

To analyze the example source code, take the file player.js version available here

1. Local variables

Local variables declaration to work with constants, SFU SDK, to display video and to work with client configuration

code

```
const constants = SFU.constants;
const sfu = SFU;
const PRELOADER_URL="../commons/media/silence.mp3";
const playStatus = "playStatus";
const playErrorInfo = "playErrorInfo";
```

2. Object to store current playback state

The object should keep Websocket session data, WebRTC connection data, room data and object to display tracks data

code

```
const CurrentState = function() {
    let state = {
        pc: null,
        session: null,
        room: null,
        display: null,
        roomEnded: false,
        set: function(pc, session, room) {
            state.pc = pc;
            state.session = session;
            state.room = room;
            state.roomEnded = false;
        },
        clear: function() {
            state.room = null;
            state.session = null;
```

```
            state.pc = null;
            state.roomEnded = false;
        },
        setRoomEnded: function() {
            state.roomEnded = true;
        },
        isRoomEnded: function() {
            return state.roomEnded;
        },
        isConnected: function() {
            return (state.session && state.session.state() ===
constants.SFU_STATE.CONNECTED);
        },
        isActive: function() {
            return (state.room && !state.roomEnded && state.pc);
        },
        setDisplay: function (display) {
            state.display = display;
        },
        disposeDisplay: function () {
            if (state.display) {
                state.display.stop();
                state.display = null;
            }
        }
    };
    return state;
}
```

3. Initialization

`init()` code

The `init()` function is called on page load and:

- initializes state objects
- initializes input fields

```
const init = function() {
    $("#playBtn").prop('disabled', true);
    $("#url").prop('disabled', true);
    $("#streamName").prop('disabled', true);
    onDisconnected(CurrentState());
    $("#url").val(setURL());
}
```

4. Establishing server connection

`RTCPeerConnection()`, `SFU.createRoom()` code

The `connect()` function is called by `Play` button click:

- creates `PeerConnection` object
- cleans the previous session state displayed
- sets up room configuration and creates Websocket session
- subscribes to Websocket session events

```
const connect = async function(state) {
    // Create peer connection
    let pc = new RTCPeerConnection();
    // Create a config to connect to SFU room
    const roomConfig = {
        // Server websocket URL
        url: $("#url").val(),
        // Use stream name as room name to play ABR
        roomName: $("#streamName").val(),
        // Make a random participant name from stream name
        nickname: "Player-" + $("#streamName").val() + "-" + createUUID(4),
        // Set room pin
        pin: 123456
    }
    // Clean state display items
    setStatus(playStatus, "");
    setStatus(playErrorInfo, "");
    try {
        // Connect to the server (room should already exist)
        const session = await sfu.createRoom(roomConfig);
        // Set up session ending events
        session.on(constants.SFU_EVENT.DISCONNECTED, function() {
            onStopClick(state);
            onDisconnected(state);
            setStatus(playStatus, "DISCONNECTED", "green");
        }).on(constants.SFU_EVENT.FAILED, function(e) {
            onStopClick(state);
            onDisconnected(state);
            setStatus(playStatus, "FAILED", "red");
            if (e.status && e.statusText) {
                setStatus(playErrorInfo, e.status + " " + e.statusText,
"red");
            } else if (e.type && e.info) {
                setStatus(playErrorInfo, e.type + ": " + e.info, "red");
            }
        });
        // Connected successfully
        onConnected(state, pc, session);
        setStatus(playStatus, "CONNECTING...", "black");
    } catch(e) {
        onDisconnected(state);
        setStatus(playStatus, "FAILED", "red");
        setStatus(playErrorInfo, e, "red");
    }
}
```

5. Playback start after session establishing

`onConnected()` code

The `onConnected()` function:

- sets up `Stop` button click actions
- subscribes to `SFU_ROOM_EVENT.PARTICIPANT_LIST` event to check if the stream is published in the room
- subscribes to room error events
- calls playback function

```
const onConnected = async function(state, pc, session) {
    state.set(pc, session, session.room());
    $("#playBtn").text("Stop").off('click').click(function () {
        onStopClick(state);
    });
    $('#url').prop('disabled', true);
    $("#streamName").prop('disabled', true);
    // Add room event handling
    state.room.on(constants.SFU_ROOM_EVENT.PARTICIPANT_LIST, function(e) {
        // If the room is empty, the stream is not published yet
        if (!e.participants || e.participants.length === 0) {
            setStatus(playErrorInfo, "ABR stream is not published", "red");
            onStopClick(state);
        }
        else {
            setStatus(playStatus, "ESTABLISHED", "green");
            $("#placeholder").hide();
        }
    }).on(constants.SFU_ROOM_EVENT.FAILED, function(e) {
        // Display error state
        setStatus(playErrorInfo, e, "red");
    }).on(constants.SFU_ROOM_EVENT.OPERATION_FAILED, function (e) {
        onOperationFailed(state);
    }).on(constants.SFU_ROOM_EVENT.ENDED, function () {
        // Publishing is stopped, dispose playback and close connection
        setStatus(playErrorInfo, "ABR stream is stopped", "red");
        state.setRoomEnded();
        onStopClick(state);
    }).on(constants.SFU_ROOM_EVENT.DROPPED, function () {
        // Client dropped from the room, dispose playback and close
connection
        setStatus(playErrorInfo, "Playback is dropped due to network issues",
"red");
        state.setRoomEnded();
        onStopClick(state);
    });
    await playStreams(state);
    // Enable button after starting playback #WCS-3635
    $("#playBtn").prop('disabled', false);
}
```

6. Streams playback

`playStreams()`, `initRemoteDisplay()`, `SFURoom.join()` code

The `playStreams()` function:

- initializes a base container tag to display incoming media streams
- sets up ABR to switch automatically between available qualities when playback channel conditions are changing
- negotiates WebRTC connection

```
const playStreams = async function (state) {
    try {
        // Create remote display item to show remote streams
        const display = initRemoteDisplay(state.room,
document.getElementById("remoteVideo"), {quality:true, autoAbr: true},
{thresholds: [
            {parameter: "nackCount", maxLeap: 10},
        {parameter: "freezeCount", maxLeap: 10},
        {parameter: "packetsLost", maxLeap: 10}
    ], abrKeepOnGoodQuality: ABR_KEEP_ON_QUALITY, abrTryForUpperQuality:
ABR_TRY_UPPER_QUALITY, interval:
ABR_QUALITY_CHECK_PERIOD},createDefaultMeetingController,
createDefaultMeetingModel, createDefaultMeetingView,
oneToOneParticipantFactory(remoteTrackProvider(state.room)));
        state.setDisplay(display);
        // Start WebRTC negotiation
        await state.room.join(state.pc, null, null, 1);
    } catch(e) {
        if (e.type === constants.SFU_ROOM_EVENT.OPERATION_FAILED) {
            onOperationFailed(state, e);
        } else {
            console.error("Failed to play streams: " + e);
            setStatus(playErrorInfo, e.name, "red");
            onStopClick(state);
        }
    }

}
```

7. Playback stopping

`CurrentState.disposeDisplay()` code

```
const stopStreams = function(state) {
    state.disposeDisplay();
}
```

8. Play click action

`onStartClick()`, `playFirstSound()`, `connect()` code

The `onStartClick()` function:

- validates input fields
- in Safari browser, calls `playFirstSound()` before playback to automatically play incoming audio
- calls `connect()` function

```
const onStartClick = function(state) {
    if (validateForm("connectionForm")) {
        $("#playBtn").prop('disabled', true);
        if (Browser().isSafariWebRTC()) {
            playFirstSound(document.getElementById("main"),
PRELOADER_URL).then(function () {
                connect(state);
            });
        } else {
            connect(state);
        }
    }
}
```

9. Stop click actions

`onStopClick()`, `Session.disconnect()` code

The `onStopClick()` function:

- stops playback
- disconects Websocket session

```
const onStopClick = async function(state) {
    stopStreams(state);
    if (state.isConnected()) {
        $("#playBtn").prop('disabled', true);
        await state.session.disconnect();
        onDisconnected(state);
    }
}
```

10. Websocket session disconnection actions

`onDisconnected()` code

The `onDisconnected()` function:

- sets up `Play` click actions
- enables `Server url` and `Stream name` fields access

```
const onDisconnected = function(state) {
    state.clear();
```

```
    $("#placeholder").show();
    $("#playBtn").text("Play").off('click').click(function () {
        onStartClick(state);
    }).prop('disabled', false);
    $('#url').prop('disabled', false);
    $("#streamName").prop('disabled', false);
}
```