オーディオダンプデータ取得方法

1. `chrome://webrtc-internals/`を開き、Create Dump 横の▶をクリック

▶ Create Dump
Read stats From: [Standardized (promise-based) getStats() API ▼]

Note: computed stats are in []. Experimental stats are marked with an * at the end and do not show up in the getStats result.

2. 下記２つのチェックを有効にする。

▼ Create Dump
[Download the PeerConnection updates and stats data]

☑ Enable diagnostic audio recordings → log audio input, audio output and echo cancellation dump

A diagnostic audio recording is used for analyzing audio problems. It consists of several files and contains the audio played out to the speaker (output) and captured from the microphone (input). The data is saved locally. Checking this box will enable recordings of all ongoing input and output audio streams (including non-WebRTC streams) and for future audio streams. When the box is unchecked or this page is closed, all ongoing recordings will be stopped and this recording functionality disabled. Recording audio from multiple tabs is supported as well as multiple recordings from the same tab.

When enabling, select a base filename to which the following suffixes will be added:

<base filename>.<render process ID>.aec_dump.<AEC dump recording ID>
<base filename>.input.<stream recording ID>.wav
<base filename>.output.<stream recording ID>.wav

It is recommended to choose a new base filename each time the feature is enabled to avoid ending up with partially overwritten or unusable audio files.

☑ Enable diagnostic packet and event recording → log event

A diagnostic packet and event recording can be used for analyzing various issues related to thread starvation, jitter buffers or bandwidth estimation. Two types of data are logged. First, incoming and outgoing RTP headers and RTCP packets are logged. These do not include any audio or video information, nor any other types of personally identifiable information (so no IP addresses or URLs). Checking this box will enable the recording for ongoing WebRTC calls and for future WebRTC calls. When the box is unchecked or this page is closed, all ongoing recordings will be stopped and this recording functionality will be disabled for future WebRTC calls. Recording in multiple tabs or multiple recordings in the same tab will cause multiple log files to be created. When enabling, a filename for the recording can be entered. The entered filename is used as a base, to which the following suffixes will be appended.

<base filename>_<date>_<timestamp>_<render process ID>_<recording ID>

If a file with the same name already exists, it will be overwritten. No more than 5 logfiles will be created, and each of them is limited to 60MB of storage. On Android these limits are 3 files of at most 10MB each. When the limit is reached, the checkbox must be unchecked and rechecked to resume logging.

3. 御社アプリをブラウザで立ち上げ、チャンネルに参加する。

Read Stats From: [Standardized (promise-based) getStats() API ▼]

Note: computed stats are in []. Experimental stats are marked with an * at the end.

| GetUserMedia Requests | https://meet.google.com/mqf-xwpt-ekn [rid: 25, lid: 1, pid: 60644] |

https://meet.google.com/mqf-xwpt-ekn, { iceServers: [], iceTransportPolicy: all, bundlePolicy: max-bundle, rtcpMuxPolicy: require, iceCandidatePoolSize: 0, sdpSemantics: "plan-b", extmapAllowMixed: true }, {advanced: [{googHighStartBitrate: {exact: 0}}, {googPayloadPadding: {exact: true}}, {googScreencastMinBitrate: {exact: 100}}, {enableDscp: {exact: true}}]}

- Plan B SDP semantics, which is used when constructing an RTCPeerConnection with {sdpSemantics:"plan-b"}, is a legacy version of the Session Description Protocol that has severe compatibility issues on modern browsers. The standardized SDP format, "unified-plan", has been used by default since M72 (January, 20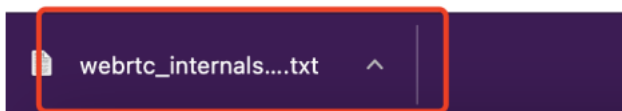19). Dropping support for Plan B is targeted for M93. See https://www.chromestatus.com/feature/5823036655665152 for more details.

Time	Event
5/28/2021, 9:59:49 PM	▶ createLocalDataChannel
5/28/2021, 9:59:49 PM	▶ createOffer
5/28/2021, 9:59:49 PM	negotiationneeded
5/28/2021, 9:59:49 PM	createOfferOnSuccess (type: "offer", 4 sections)
5/28/2021, 9:59:50 PM	▶ setLocalDescription (munged) (type: "offer", 4 sections)
5/28/2021, 9:59:50 PM	setLocalDescriptionOnSuccess
5/28/2021, 9:59:50 PM	▶ signalingstatechange
5/28/2021, 9:59:50 PM	▶ setRemoteDescription (type: "answer", 4 sections)
5/28/2021, 9:59:50 PM	▶ icegatheringstatechange
5/28/2021, 9:59:50 PM	setRemoteDescriptionOnSuccess
5/28/2021, 9:59:50 PM	▶ signalingstatechange

4. しばらくチャンネル接続し、「Create Dump」を展開する。

5. 「Download the PeerConnection updates and stats data」をクリック

6. ダウンロードファイルを取得する。

取得したファイルを弊社まで送付いただけますと幸いです。
ご協力よろしくお願い致します。