HTML

H5STREAM

linkingvison

内网直播

WEBSOCKET/WEBRTC 采用HTML5 原生播放技术,延迟可以控制在1秒以内

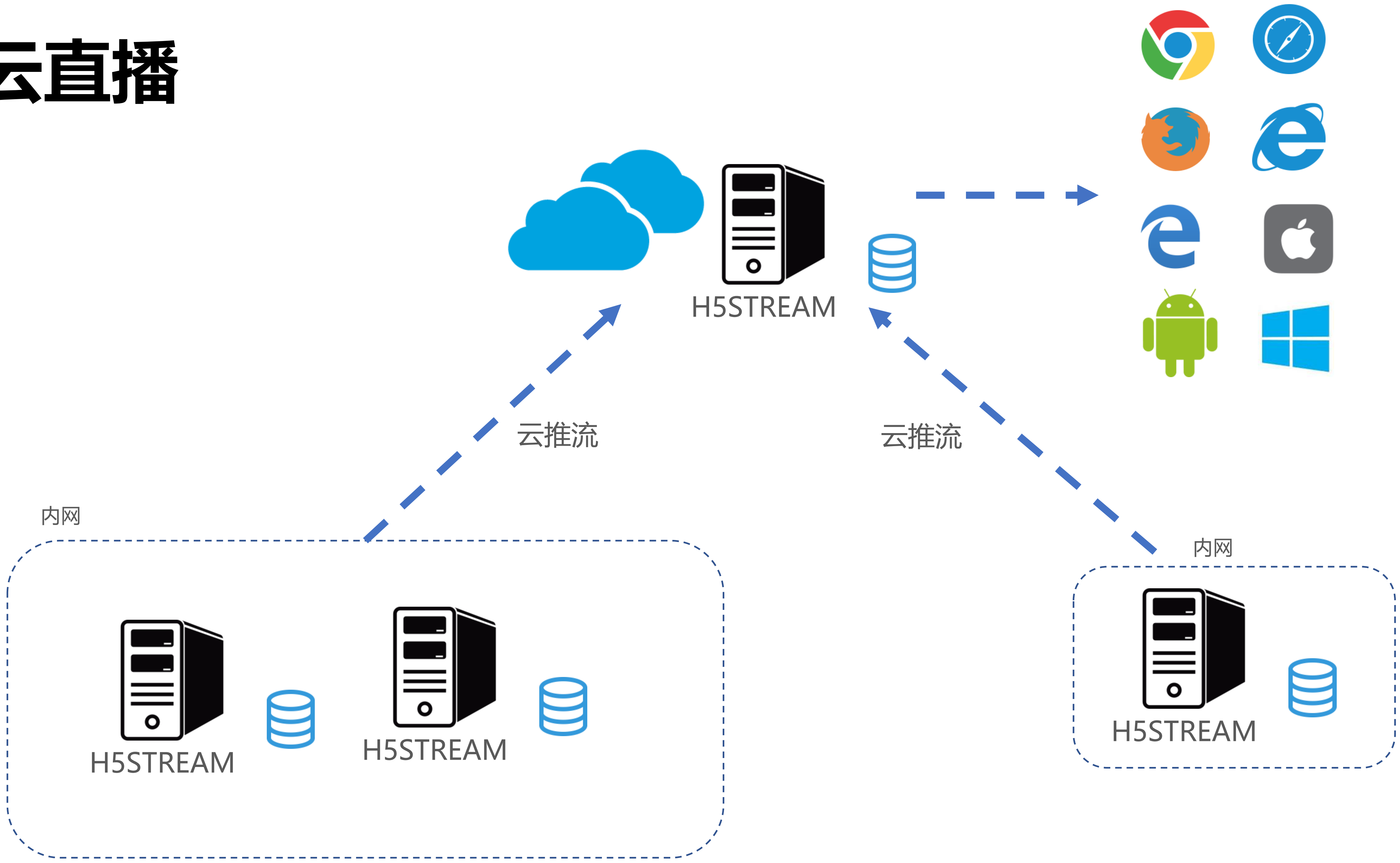
视频延迟可以达到400毫秒

所有视频都被加密

浏览器原生硬件解码渲染支持

不需要转码

云直播



RESTFUL 接口

系统

- Login
- Logout
- Keepalive

视频源管理

- GetSrc
- GetImage
- AddSrcFile
- AddSrcRTSP
- AddSrcONVIF
- DelSrc
- Ptz

ONVIF

- OnvifSearch
- OnvifProbe

录像和快照

- Record
- Snapshot
- Search

JAVASCRIPT 接口

WEBSOCKET

H5sPlayerWS

```
/**
* Interface with h5s websocket player API
* @constructor
*/

function H5sPlayerWS(conf)

H5sPlayerWS.prototype.connect
H5sPlayerWS.prototype.disconnect
```

WEBRTC

H5sPlayerRTC

```
/**
* Interface with h5s WebRTC player API
* @constructor
* @param {string} videoId - id of the
video element tag
*/
function H5sPlayerRTC(conf)

H5sPlayerRTC.prototype.connect
H5sPlayerRTC.prototype.disconnect
```

RTMP

VideoJS

www/rtmp.html

HLS

H5sPlayerHls

```
/**
* Interface with h5s websocket player API
* @constructor
*/
function H5sPlayerHls(conf)
H5sPlayerHls.prototype.connect
H5sPlayerHls.prototype.disconnect
```

```
/**
 @param
 var conf = {
          videoid:'h5sVideo1', //{string} - id of the video element tag
          videodom: h5svideodom1, //{object} - video dom. if there has videoid, just use the videoid
          protocol: window.location.protocol, // {string} - http: or https:
          host: window.location.host, //{string} - localhost:8080
          rootpath:window.location.pathname, // {string} - path of the app running
          token:'token1', // {string} - token of stream
          hlsver:'v1’, //{string} -  v1 is for ts, v2 is for fmp4
          session:'c1782caf-b670-42d8-ba90-2244d0b0ee83' //{string} - session got from login
*/
```

浏览器兼容性

	Chrome	Firefox	IE11	Edge	Safari	WeChat
WIN7	WEBRTC WEBSOCKET	WEBRTC WEBSOCKET	RTMP	-	-	-
WIN 8/10	WEBRTC WEBSOCKET	WEBRTC WEBSOCKET	RTMP WEBSCOKET	WEBRTC	-	-
macOS	WEBRTC WEBSOCKET	WEBRTC WEBSOCKET	-	-	WEBRTC WEBSOCKET	-
iOS 11	HLS WEBRTC	HLS WEBRTC	-	-	HLS WEBRTC	HLS WEBRTC
iOS 8-10	HLS	HLS	-	-	HLS	HLS
Android	WEBSOCKET WEBRTC	WEBRTC WEBSOCKET	-	-	-	HLS

单协议客户端

http://localhost:8080/ws.html?token=token1&autoplay=1
http://localhost:8080/rtc.html?token=token1&autoplay=1

http://localhost:8080/rtmp.html?token=token2
http://localhost:8080/hls.html?token=token2

Websocket	WebRTC	RTMP	HLS
ws.html	rtc.html	rtmp.html	hls.html

Chrome support this http://localhost:8080/rtc.html?token=token1&autoplay=1
but when change to websocket, chrome does't allow
Uncaught (in promise) DOMException: play() failed because the user didn't interact with the document first. https://goo.gl/xX8pDD

配置文件

conf/h5ss.conf	
HTTP	HTTP HTTPS 服务器配置
RTSP	RTSP 服务器配置, SSL 代表 RTSP over TCP/TLS
RTMP	RTMP服务器配置, SSL 代表 RTMP over TCP/TLS
FLV	FLV服务器配置, SSL 代表 FLV over HTTPS
HLS	HLS服务器配置，包括HLS版本及参数配置
WEBRTC	WEBRTC 配置
SYSTEM	H5stream系统配置，包括日志和线程池配置
USER	用户管理配置
SOURCE	视频源配置，包括文件 RTSP/RTMP/ONVIF

音频开启

h5stream 从5.4开始支持AAC 音频，已经测试过HIKVISION的网络摄像机。
从6.0 开始，h5stream 对G711/PCM 音频进行转码成AAC以便支持音频

1.在配置文件中开启音频支持


```
"strPasswd": "admin12345",
"bPasswdEncryptComment": "Password Encrypted",
"bPasswdEncrypt": false,
"bEnableAudioComment": "Enable Audio",
"bEnableAudio": true,
"nConnectTypeComment": "H5_ONDEMAND/H5_ALWAYS/H5_AUTO",
"nConnectType": "H5_AUTO",
"nRTSPTypeComment": "RTSP Connect protocol H5_RTSP_TCP/H5_
"nRTSPType": "H5_RTSP_AUTO",
"strSrcIpAddressComment": "Ip Address for the device",
"strSrcIpAddress": "192.168.100.173",
```


2.把网络摄像机的音频格式修改为AAC