

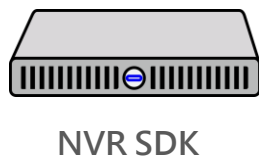
**HTML**



**H5STREAM**

Copyright © 2018 linkingvison, All rights reserved

# ON-PREMISES



REST API



H5STREAM

RECORD SNAPSHOT



RTSP/RTMP

HLS

WEBSOCKET

WEBRTC



WEBSOCKET/WEBRTC IS HTML5  
NATIVE PLAYER WITH LOW  
LATENCY(<1s)

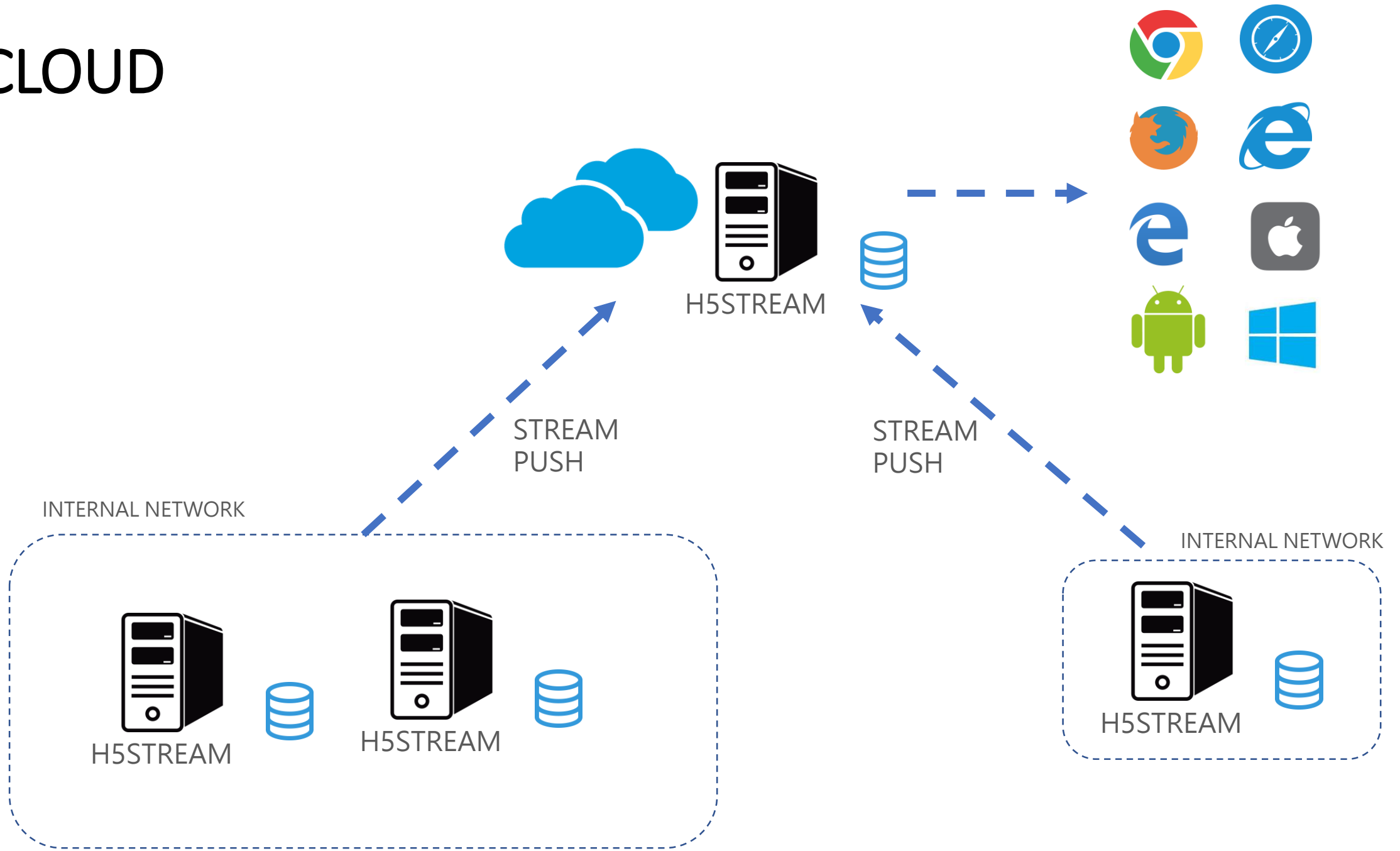
THE LATENCY CAN BE 400ms

ALL VIDEO IS ENCRYPTED

HARDWARE DECODE/RENDERING IN BROWSER

NO TRANSCODING

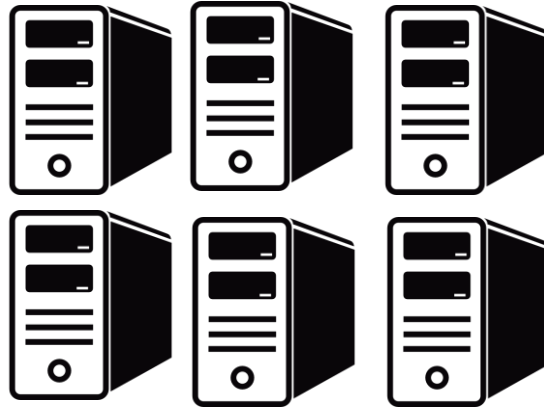
# CLOUD



# CLUSTER



Consul based  
microservice, auto  
service register



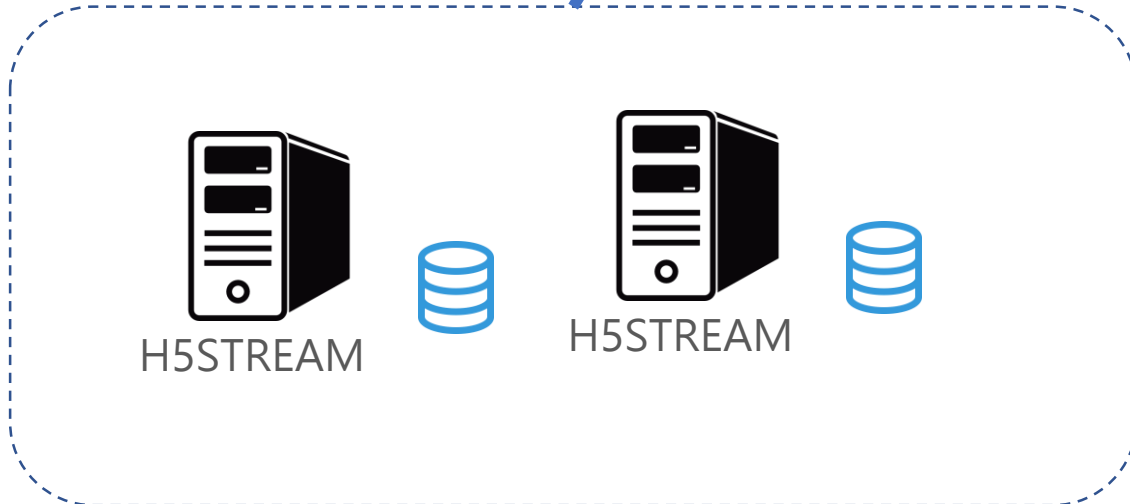
H5STREAM

AUTO SELECT  
SERVICE PLAY



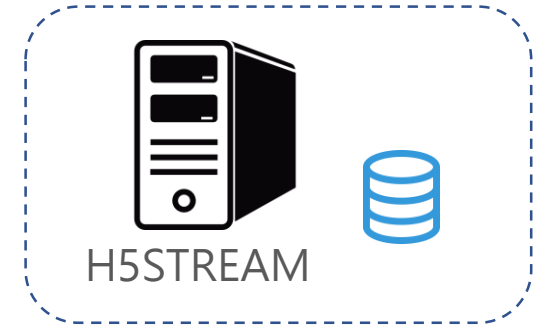
AUTO SELECT  
SERVICE STREAM PUSH

INTERNAL NETWORK



AUTO SELECT  
SERVICE STREAM PUSH

INTERNAL NETWORK



# RESTFUL API

## SYSTEM

Login

Logout

Keepalive

## SOURCE MANAGEMENT

GetSrc

GetImage

AddSrcFile

AddSrcRTSP

AddSrcONVIF

DelSrc

Ptz

## ONVIF

OnvifSearch

OnvifProbe

## RECORD

Record

Snapshot

Search

# JAVASCRIPT API

## 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
 */
function H5sPlayerRTC(conf)
```

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

## RTMP

### VideoJS

[www/rtmp.html](http://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;
 }
 */
```

# COMPATIBILITY

	Chrome	Firefox	IE11	Edge	Safari	WeChat
WIN7	WEBRTC WEBSOCKET	WEBRTC WEBSOCKET	RTMP	-	-	-
WIN 8/10	WEBRTC WEBSOCKET	WEBRTC WEBSOCKET	RTMP WEBSOCKET	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	-	-	-	WEBSOCKET

# CONFIGURATION FILE

conf/h5ss.conf	
HTTP	HTTP HTTPS server configuration
RTSP	RTSP server configuration, SSL is RTSP over TCP/TLS
RTMP	RTMP server configuration, SSL is RTMP over TCP/TLS
FLV	FLV server configuration, SSL is FLV over HTTPS
HLS	HLS version and parameter configuration
WEBRTC	WEBRTC configuration
SYSTEM	H5stream system configuration such as log and HTTP server thread
USER	User management configuration
SOURCE	Video source configuration, include File/RTSP/RTMP/ONVIF



# PROTOCOL CLIENT

<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 doesn't allow

Uncaught (in promise) DOMException: play() failed because the user didn't interact with the document first. <https://goo.gl/xX8pDD>


# AUDIO ENABLE

h5stream from 5.4 version support AAC audio, has tested with HIKVISION camera.  
After 6.0 G711/PCM will auto to decode and encode to AAC

1. Enable the audio from configuration, API also support enable when added.

```
...
"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. Change the camera audio format to AAC or G711A G711U

视频	音频	ROI	码流信息叠加	区域裁剪
音频编码	AAC			
采样率	16kHz			
音频码率	32kbps			
音频输入	LineIn			
输入音量	 19			
环境噪声过滤	关闭			