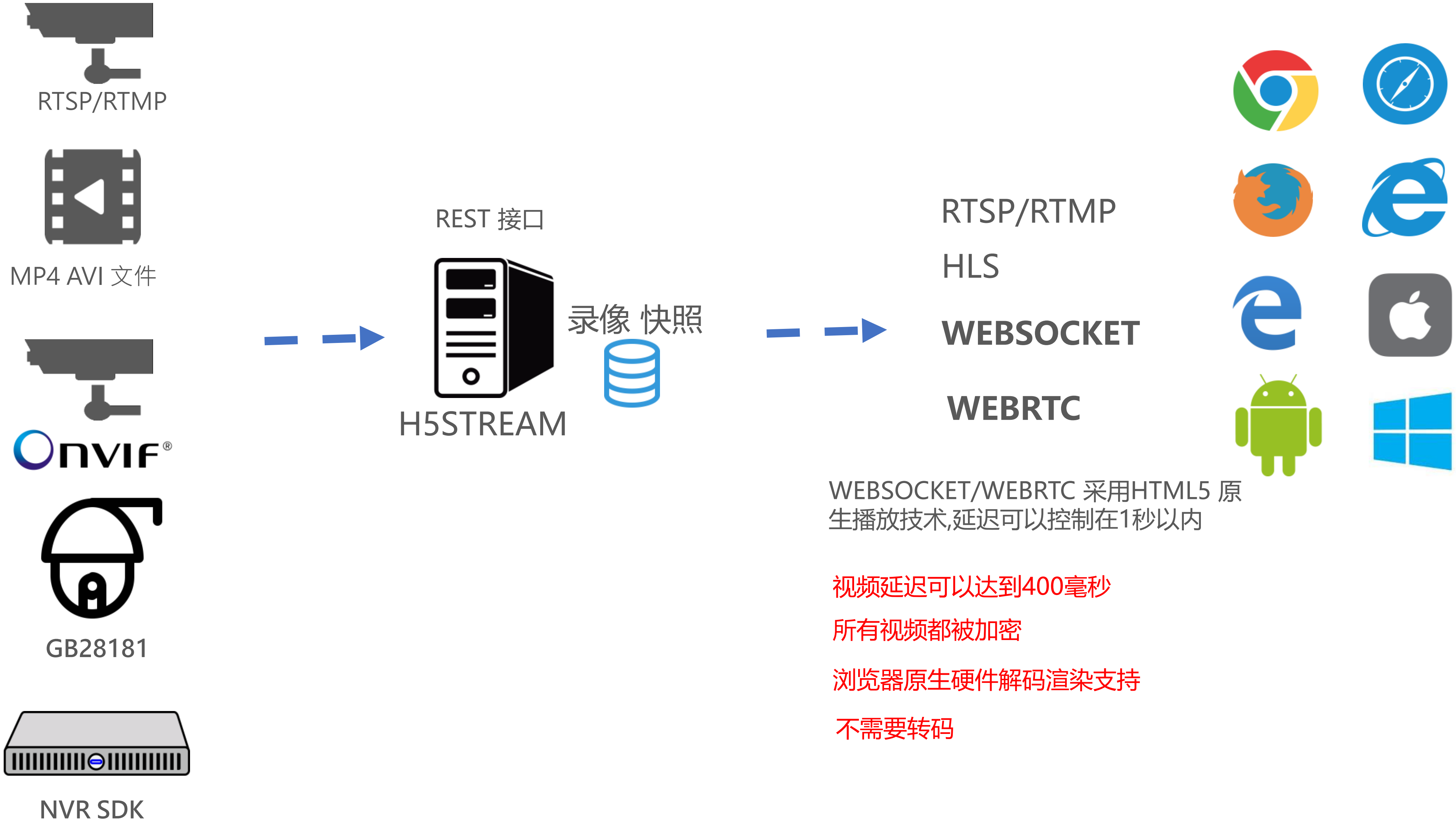
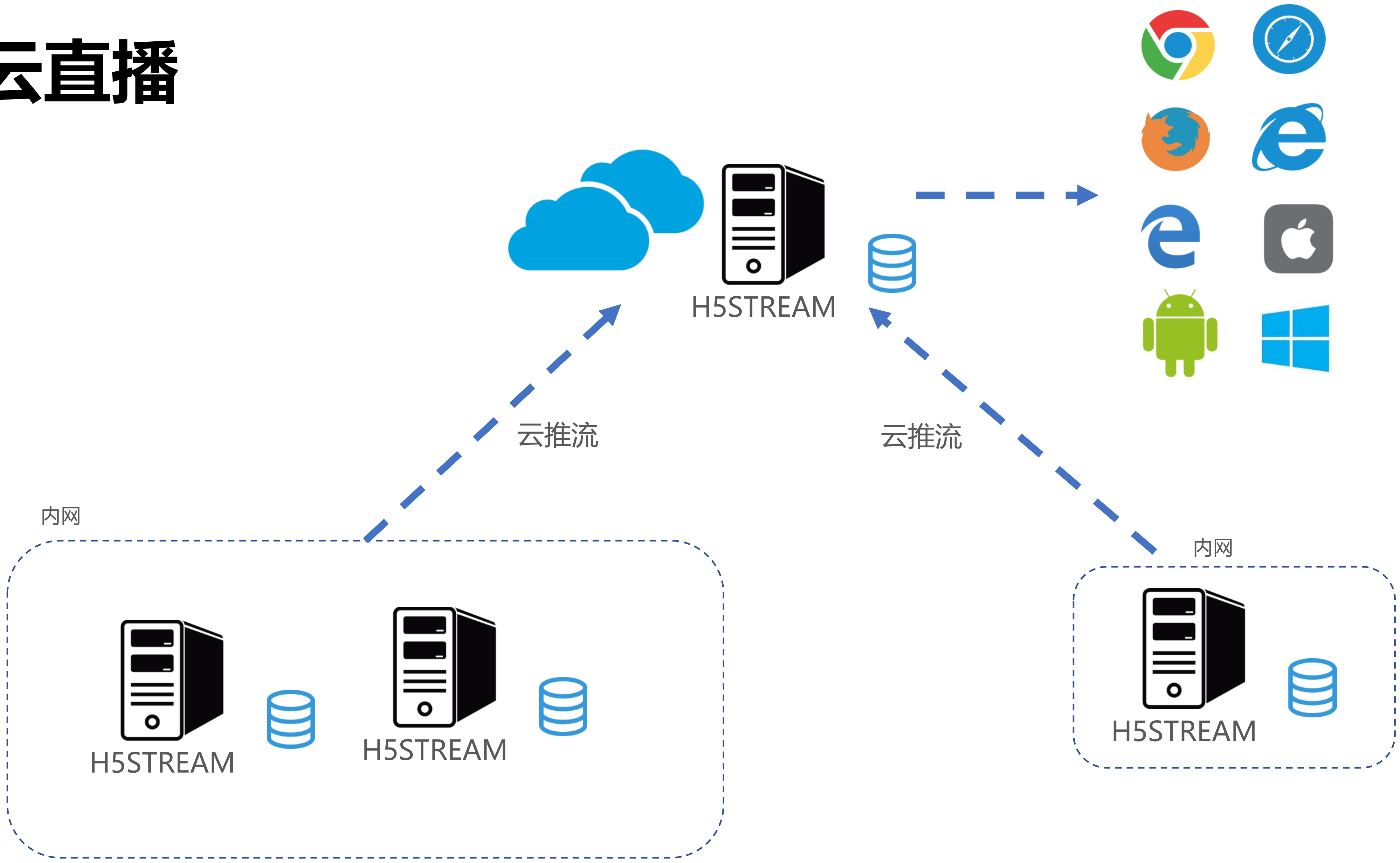
H5STREAM

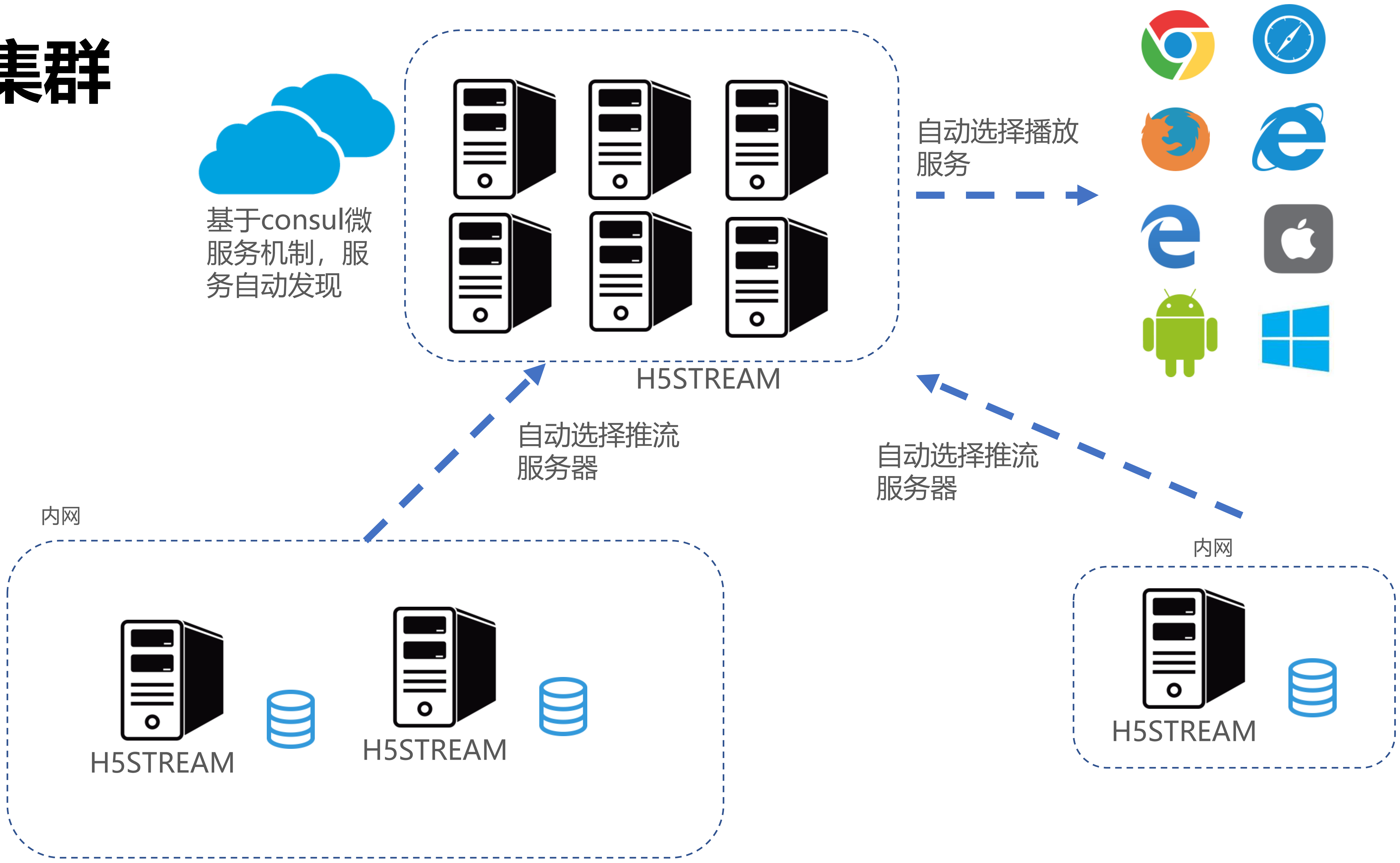
内网直播


RTSP/RTMP
MP4 AVI 文件
ONVIF®
GB28181
NVR SDK
REST 接口
录像 快照
H5STREAM
RTSP/RTMP
HLS
WEBSOCKET
WEBRTC
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



集群


基于consul微服务机制，服务自动发现
H5STREAM
自动选择播放服务
自动选择推流服务器
自动选择推流服务器
内网
H5STREAM
H5STREAM
内网
H5STREAM

RESTFUL 接口

SYSTEM	SOURCE & DEVICE MANAGEMENT	ONVIF	RECORD	CLOUD	CLUSTER
Login	GetSrc	OnvifSearch	Record	GetCloudClientInfo	GetServiceAddr
Logout	GetImage	OnvifProbe	Snapshot	GetServerList	GetClusterStatus
Keepalive	AddSrcFile		Search		
	AddSrcRTSP				
	AddSrcONVIF				
	DelSrc				
	Ptz				
	AddDeviceHik				
	AddDeviceDh				

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
*/
function H5sPlayerRTC(conf)

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

RTMP

VideoJS

www/rtmp.html

Flowplayer

www/rtmp2.html

HLS

H5sPlayerHls

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

```
/**
 * @constructor
 * @param
 var pbconf1 = {
            begintime: '2019-03-23T120101+08',//{string} begintime
            endtime: '2019-03-23T150101+08',//{string} endtime
            callback: PlaybackCB, //{function}(event(string), userdata(object))
            userdata:  user data // user data
};

 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
            pbconf: pbconf1, //This is optional, if no pbconf, this will be live.
            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	-	-	-	WEBSOCKET

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
CLOUD	H5stream 云模式配置
CLUSTER	H5stream 集群配置
GB28181	GB28181 服务配置
USER	用户管理配置
SOURCE	视频源配置，包括文件 RTSP/RTMP/ONVIF
DEVICE	设备SDK 接入配置，包含HIKVISION/DAHUA

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

如果想在Chrome支持开启播放功能，可以参考FAQ

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，API也支持在添加时候开启音频


```
"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",
```



```
"dev": [
{
"strNameComment": "name for this device",
"strName": "Device 1",
"strTokenComment": "token for this device, must unique, if same, only first will be available",
"strToken": "device1",
"nTypeComment": "device type H5_DEV_H5S/H5_DEV_ONVIF/H5_DEV_HIK/H5_DEV_DH...",
"nType": "H5_DEV_HIK",
"strUserComment": "username",
"strUser": "admin",
"strPasswdComment": "password",
"strPasswd": "12345",
"bPasswdEncryptComment": "Password Encrypted",
"bPasswdEncrypt": false,
"strDevIpAddressComment": "Ip Address for the device",
"strDevIpAddress": "192.168.0.1",
"strDevPortComment": "Port for the device",
"strDevPort": "8000",
"bEnableAudioComment": "Enable Audio",
"bEnableAudio": false
```


2.把网络摄像机的音频格式修改为AAC，或者G711A G711U