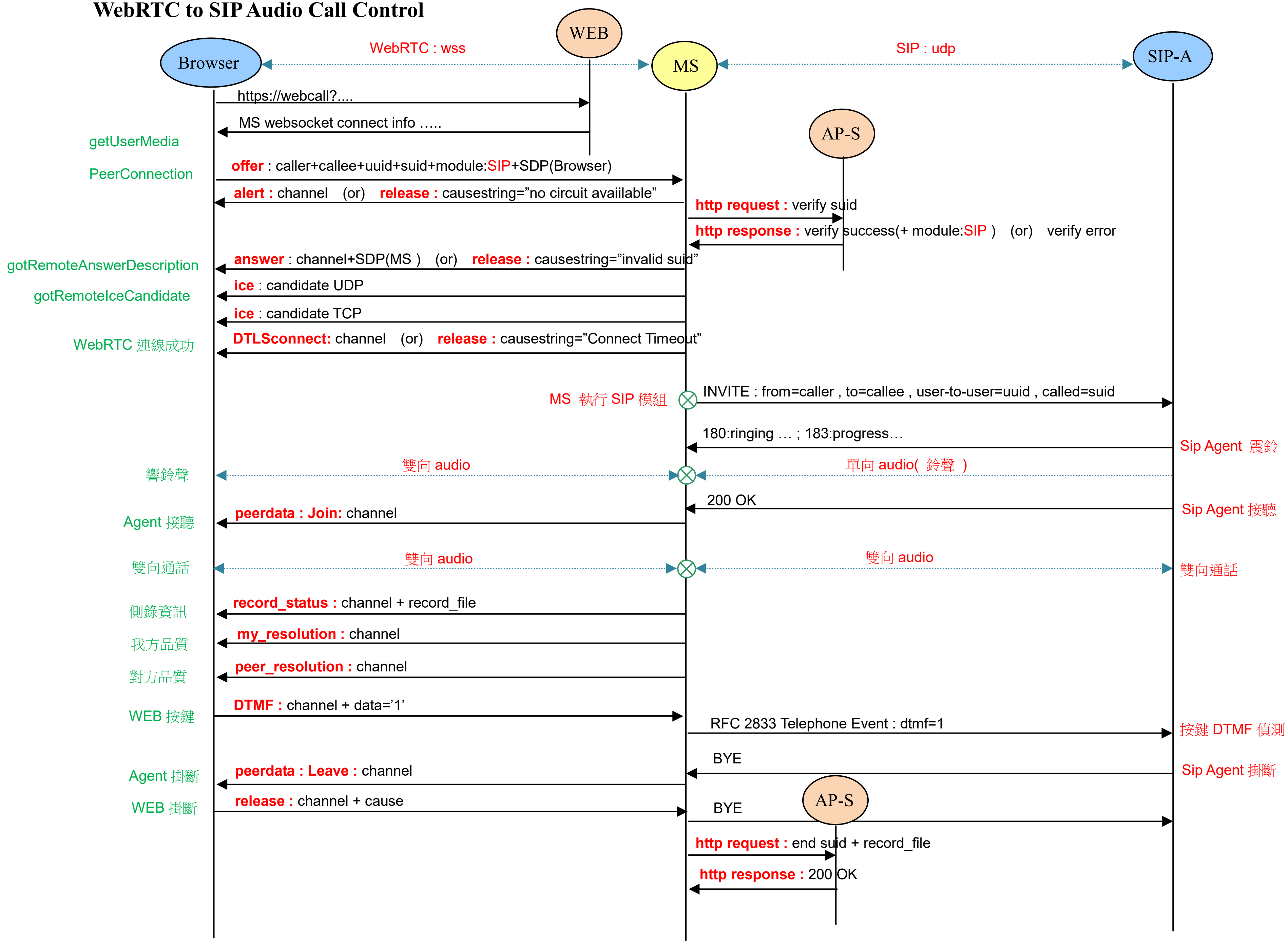
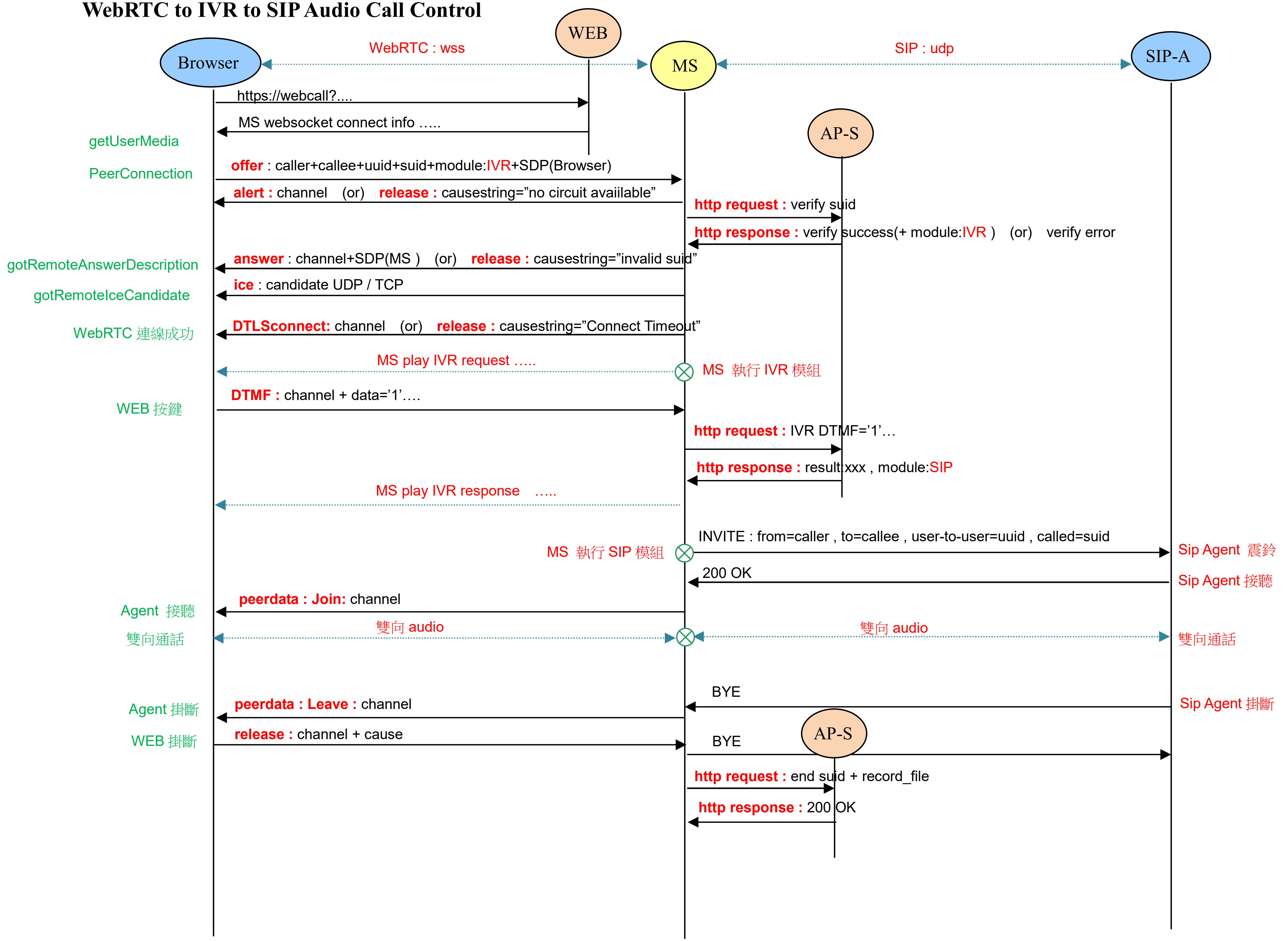
WebRTC to SIP Audio Call Control



WebRTC to IVR to SIP Audio Call Control



Action List : **(JSON format - Key:Value)** **[註]:** 長度限制**: Key <=16 bytes ; Value <=128 bytes**

Action	功能	參數(Mandatory)	參數(Optional)	方向
offer	Browser 建立 WebRTC 連線	dialogid : dialog reference number (UINT) ref : call reference number (UINT)，配合 dialogid 必須唯一 medium : media type : 0=audio 2=video suid : session unique id (CHAR[64]) ua_family : client 端 UA 瀏覽器 (CHAR[64]) ua_version : client 端 UA 版本(CHAR[64]) ua_os : client 端 OS (CHAR[64]) SDP: { Browser SDP 參數集合 }	caller : 主叫號碼 (UINT) callee : 被叫號碼 (UINT) uuid : user to user information (CHAR[128]) roomkey : 會議室鑰匙號碼(UINT) record : 1:啟動側錄 0:no(default) nickname : 暱稱(CHAR[128]) module : MS 進線模組名稱(CHAR[6]) lifetime : 最長接通秒數(UINT),0:不限制 audio : 聲音格式 PCMU,OPUS protocol: UDP(default) or TCP for RTP	To MS

{"action":"offer","dialogid":13290721,"ref":2392510,"record":1,"medium":0,"caller":5678,"suid":"20200406061215567893333","roomkey":6655,”module”:”SIP”,
"callee":9333,"uuid":"TEST","lifetime":600,"ua_family":"Chrome","ua_version":"80","ua_os":"Windows 10","location_ip":"rtc.tw","audio":"PCMU",
"offer":{"type":"offer","sdp":"v=0\r\n…………}”

[SIP Message] :

Via: SIP/2.0/UDP 10.140.0.2;rport;branch=z9hG4bK3NccFjavmH1ZD

Max-Forwards: 70

From: "5678" <sip:5678@35.221.193.150>;tag=vFNec197SaU3S

To: "9333" <sip:9333@118.166.13.212:43894>

Call-ID: 20200406061215567893333

CSeq: 2 INVITE

Contact: <sip:5678@35.221.193.150:5060>

User-Agent: WebRTC Media Server 5.10

User-to-User: 54455354;encoding=hex

<table>
<tr><td>alert</td><td>MS 取得空餘 channel
執行這通 offer request.</td><td>dialogid + ref： 整通 call 過程必須維持此二參數
channel : MS channel number
[註] 如果 MS 沒有空餘線路,回應如下:
{"action":"release","channel":0,"cause":34,"causestring":"no circuit available"}</td><td></td><td>From MS</td></tr>
<tr><td colspan="5">{"action":"alert","dialogid":13290721,"ref":2392510,"channel":"4",}</td></tr>
</table>

<table>
<tr><td>answer</td><td>MS 建立 WebRTC 連線</td><td>dialogid + ref： 整通 call 過程必須維持此二參數
channel : MS channel number
SDP : { MS SDP 參數集合 }</td><td></td><td>From MS</td></tr>
<tr><td colspan="5">{"action":"answer","dialogid":13290721,"ref":2392510,"channel":"4",,"medium":0,"answer":{"type":"answer","sdp":"v=0\r\................}”</td></tr>
</table>

<table>
<tr><td>ice</td><td>MS 的 ICE for RTP address</td><td>dialogid + ref： 整通 call 過程必須維持此二參數
channel : MS channel number
ice : candidate UDP , TCP
ICE 有 UDP,TCP 兩種;通常以 UDP 為優先，如果遇到 UDP 不通時會自行轉換為 TCP</td><td></td><td>From MS</td></tr>
<tr><td colspan="5">{"action":"ice","dialogid":13290721,"ref":2392510,"channel":"4","ice":{"candidate":"candidate:1 1 udp 2113929727 35.221.193.150 30003 typ host generation 0 ufrag IiCTnyfJQsA6suXc","sdpMid":"0","sdpMLineIndex":0}}
{"action":"ice","dialogid":13290721,"ref":2392510,"channel":"4","ice":{"sdpMid":"0","sdpMLineIndex":0,"candidate":"candidate:2 1 tcp 2113917382 35.221.193.150 1701 typ host generation 0 ufrag IiCTnyfJQsA6suXc"}}</td></tr>
</table>

<table>
<tr><td>DTLSconnect</td><td>DTLS handshake success</td><td>dialogid + ref： 整通 call 過程必須維持此二參數
channel : MS channel number</td><td></td><td>From MS</td></tr>
<tr><td colspan="5">{"action":"DTLSconnect","dialogid":13290721,"ref":2392510,"channel":"4","medium":"audio"}
下面是 DTLS handshake 失敗訊息 ：
{"action":"release","dialogid":13290721,"ref":2392510,"channel":"4","cause":"16","causestring":"Connect Timeout"}</td></tr>
</table>

release	中斷 WebRTC 連線	dialogid + ref： 整通 call 過程必須維持此二參數 channel : MS channel number	cause : release cause number,eg. 16:normal causestring : cause description	Both way
{"action":"release","dialogid":13290721,"ref":2392510,"channel":"4","cause":"16","causestring":"normal release"}				

record_status	啟動側錄時,MS 通知側錄檔狀態	dialogid + ref： 整通 call 過程必須維持此二參數 channel : MS channel number medium : media type : 0=audio 2=video status : “start”表示開始側錄 record_file： 側錄檔名		From MS
{"action":"record_status","dialogid":13290721,"ref":2392510,"status":"start","channel":"4","medium":"0","record_file":"audio/20200405/13290723392510_004.wav"}				

my_resolution	我的聲音/影像品質	dialogid + ref： 整通 call 過程必須維持此二參數 channel : MS channel number width : video 寬 height : video 長 REMB : Video Bit-Rate 期待值 framerate : video 每秒 frame 數(正常~30) audio_bitrate： 聲音頻寬(K-bits) video_bitrate： 影像頻寬(K-bits) jitter： 聲音封包抖動時間(ms) packetloss： 封包遺失筆數		From MS
{"action":"my_resolution","dialogid":13290721,"ref":2392510,"width":0,"height":0,"video_bitrate":0,"REMB":500,"framerate":0,"audio_bitrate":20, "jitter":20,"packetloss":0,"channel":4}				

<table>
<tr><td>peer_resolution</td><td>對方的聲音/影像品質</td><td>dialogid + ref：整通 call 過程必須維持此二參數
channel：對方的 MS channel number
width : video 寬
height : video 長
REMB : Video Bit-Rate 期待值
framerate : video 每秒 frame 數(正常~30)
audio_bitrate：聲音頻寬(K-bits)
video_bitrate：影像頻寬(K-bits)
jitter：聲音封包抖動時間(ms)
packetloss：封包遺失筆數</td><td></td><td>From MS</td></tr>
<tr><td colspan="5">{"action":"peer_resolution","dialogid":13290721,"ref":2392510,"width":0,"height":0,"video_bitrate":0,"REMB":0,"framerate":0,"audio_bitrate":63,
"jitter":58,"packetloss":3,"channel":66}</td></tr>
</table>

<table>
<tr><td>peerdata</td><td>APP 之間的資料傳輸，或 MS 通知 APP, 其他 peer 的進線(Join)或離線(Leave)等訊息</td><td>dialogid + ref：整通 call 過程必須維持此二參數
channel： MS channel number of mine
peerchannel : MS channel number of peer ,
type : “indication” or “command”
medium : “audio” or “video” – peer 的 medium 屬性
peerdata: data content</td><td>nickname : sender 暱稱
userid : sender userid</td><td>Both Way</td></tr>
<tr><td colspan="5">{"action":"peerdata","dialogid":13290721,"ref":2392510,"channel":"4","peerchannel":"66","medium":"audio","userid":"","nickname":"","type":"indication",
"peerdata":"Join","value":"width=0000;height=0000"}
{"action":"peerdata","dialogid":13290721,"ref":2392510,"channel":"4","peerchannel":"66","medium":"audio","userid":"","nickname":"","type":"indication",
"peerdata":"Leave",”value”:”width=0000;height=0000"}</td></tr>
</table>

<table>
<tr><td>message</td><td>系統提示用戶的訊息,client 應該根據 level 及 display 適當顯示,在 displaytime 之後自動消失 .</td><td>dialogid + ref : 整通 call 過程必須維持此二參數
channel : MS channel number
message : message of system to notify client
level : “info”,”notify”,”warning”,”error”
position : ”top”,”bottom”,” middle”
displaytime : in seconds , disappear when timeout</td><td></td><td>From MS</td></tr>
<tr><td colspan="5">{"action":"message","dialogid":149931,"ref":14932,"channel":"5","message":"501 系統偵測到您的影像雍塞,可能是頻寬不足,建議您暫時關閉影像”,”level":”warning”,”position":”top”,”displaytime”:5}

------- System Message ID --------------
message: 501 “系統偵測到您網路延遲,暫時關閉您的影像.”
message: 502 “系統偵測到對方的網路延遲,暫時關閉對方的影像.”
message: 503 “系統偵測到您的影像壅塞,暫時關閉您的影像.”
message: 504 “系統偵測到對方的影像壅塞,暫時關閉對方的影像.”
message: 505 “系統偵測到您的網路壅塞,暫時關閉對方的影像.”
message: 506 “系統偵測到對方的網路壅塞,暫時關閉對方的影像.”
message: 510 “系統沒有收到麥克風的聲音,請關閉瀏覽器,再試一次.”</td></tr>
</table>

<table>
<tr><td>DTMF</td><td>WEB 按鍵</td><td>dialogid + ref : 整通 call 過程必須維持此二參數
channel : MS channel number
value : 按鍵值 ‘0’ ~ ‘9’ , ‘*’ , ‘#’</td><td></td><td>Both Way</td></tr>
<tr><td colspan="5">{ 'action': 'DTMF' "dialogid":149931,"ref":14932,"channel":"5", 'value': '1' }</td></tr>
</table>