

## SIP stability test data collect and analyze

This guild is to help you to have a fully test with SIP divert.

### 1. SIP work model and stability factor

As Figrue.1 shows a simple model for how SIP works, and some factors might has influence to the stability of SIP calling.

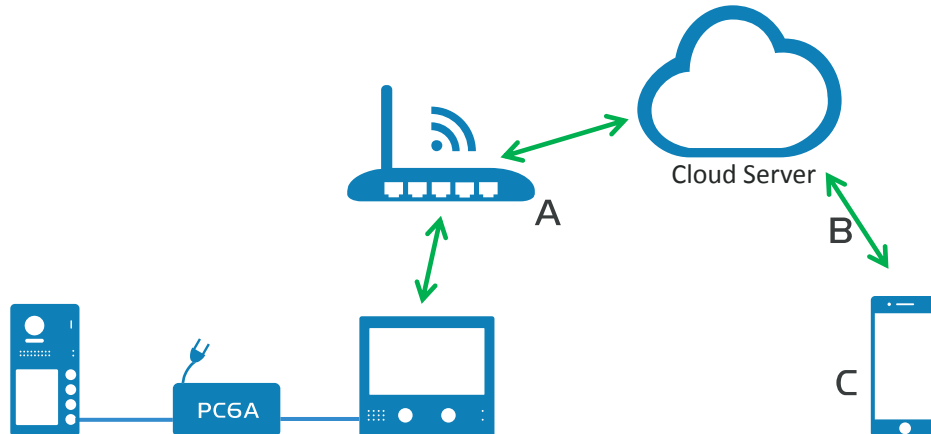


Figure1. A simple model of how SIP works

3 factors relate to the stability

- A. The quality of the Wi-Fi router connected with DX monitor, and the internet connection to Cloud Server.
- B. The quality of the Wi-Fi router connected with mobile, and the internet connection between Router and Cloud Server. Or internet connection between mobile and Cloud Server when using 4G GSM.
- C. Mobile models, OS version and APP version.

In any case of bad connection, you should always consider above 3 factors.

### 2. Process of how SIP works and diagnose log

Page 2, figure2 shows process how SIP works to get video, first found and connect to target mobile then apply for video. DX monitor in version V1.5.0 is able to collect diagnose log, a diagnose log contains below text:

Text fields	Explain
Start:820b31352b26001	Test start, with SIP account 820b31352b26001
APP Restart	APP
Recv SIP cmd[x]	Technical information only
RelInvite 1	Counts that device try to find mobile
Target Ring	Target mobile receives call and ring
Target Not found	Target mobile is offline, or can
Target Answer	Target mobile answer the call
Video connected	Video successfully shows on mobile
ReAnswer1	Counts that device try to get video when no video
Target No answer	Target mobile is not answer the call
Target Cancel	Target mobile hangs up call after pick
End: talkover	SIP call end with right process

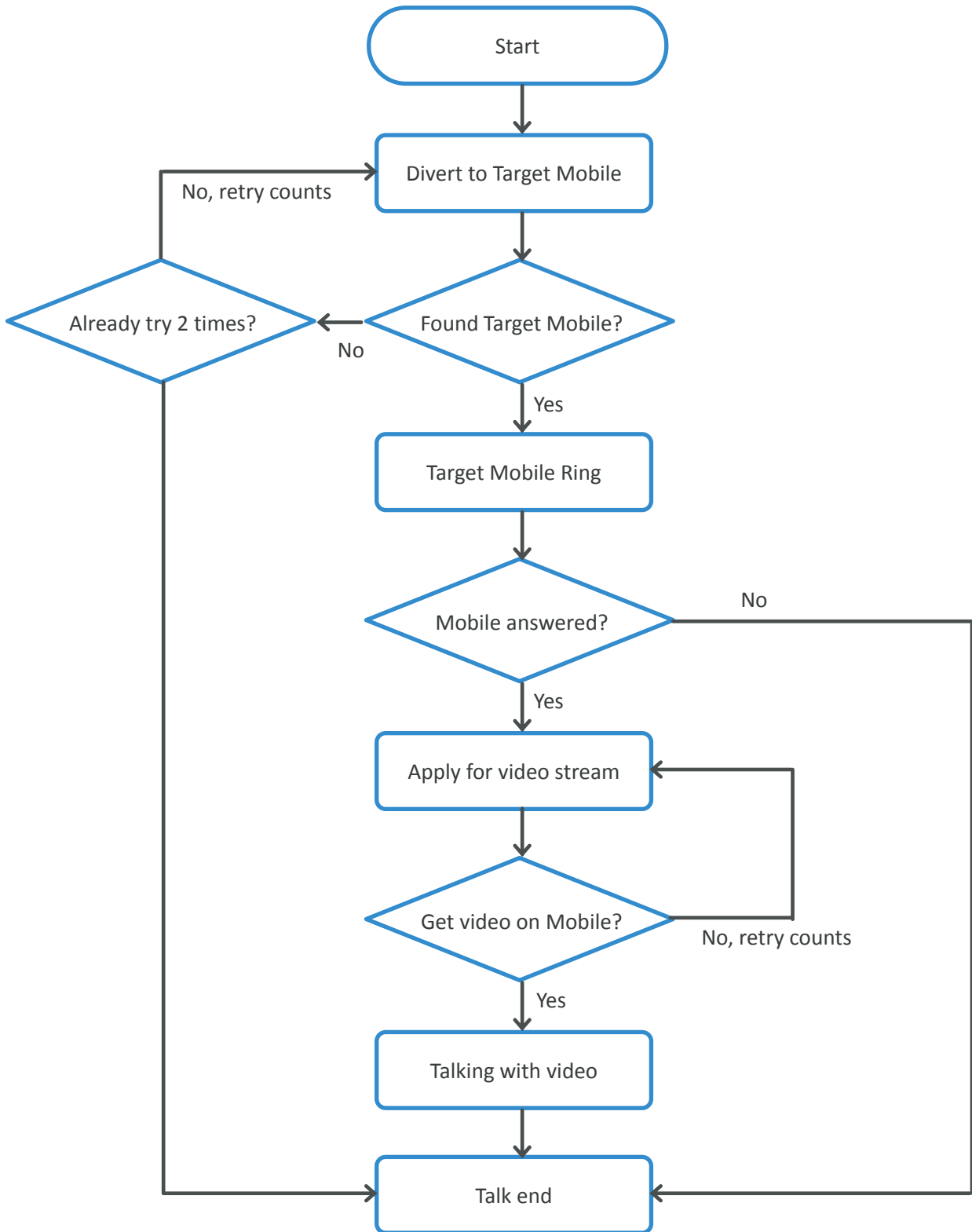


Figure2. A simple model of how SIP works

The diagnose log will also shows a conclusion of each calling with below message:

**Result: TXXX, Call(S/F, 0~2), -/Talk(S/F, 0~9)**

Means in TXXX calling, call process (successful/fail, try 0~2 times), Video Talks (successful/fail, try 0~2 times).

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As figure3 shows a part of diagnose log, with an intact calling log

TIME	Count	LOG
181020_14:51:33	T0005	Start:0e001a2b9160
181020_14:51:38	T0005	Recv_SIP_cmd[12]
181020_14:51:38	T0005	Recv_SIP_cmd[8]
181020_14:51:40	T0005	Recv_SIP_cmd[9]
181020_14:51:40	T0005	Target_Ring
181020_14:51:42	T0005	Recv_SIP_cmd[10]
181020_14:51:42	T0005	Target_Answer
181020_14:51:45	T0005	Recv_SIP_cmd[8]
181020_14:51:47	T0005	Recv_SIP_cmd[10]
181020_14:51:47	T0005	Video_connected
181020_14:52:10	T0005	Recv_SIP_cmd[25]
181020_14:52:10	T0005	Target_Cancel
181020_14:52:10	T0005	Result:T6,Call(S,0),Talk(S,0)

Events time

6th calling, SIP call and video transfer successful without any retry

Mobile ring successfull

Mobile Answered call

Video shows on Mobile

Mobile hangs up

Figure3. Part of diagnose log

## 3. SIP diagnose statistics on DX monitor

DX monitor with version V1.5.0 also provide a simple diagnose datasheet view, on the menu, "Settings" -> "Wireless" -> "SIP tools" -> "SIP call statistics", statistics will only create after a certain time of "SIP call testing" ([refers ..](#)), call statistics as below:

Call Statistics	
SIP Calls	[31/37] [Calls Divert successfully/Total Calls]
Call service	[83.8%, 56.8%] [Call success rate/Rate without retry]
SIP video	[19/25] [Calls with video stream/Total Calls]
Video service	[76.0%, 68.0%] [Video success rate/Rate without retry]
SIP server delay	[15s, 66ms] [SIP calling response timing/Delay]
In testing, 49 remaining, waiting [Testing status]	

Figure4. Call Statistics

SIP server delay [15s, 66ms]

To check the connection quality from DX monitor to Server, below:

First number show an average response time of latest 10 calls, second number shows "Ping" delay to server.

## 4. Typical SIP calling test procedure

Below is a general SIP calling test to see if your network or DX monitor is satisfied to the SIP calling.

1) Start automatic SIP divert test:

On DX monitor, tap “Settings” -> “Wireless” -> “SIP tools” -> “SIP call testing”. In this page:

- ✧ “SIP call testing count” from 1 to 99, means test automatically 1-99 times;
- ✧ Set “Call interval” from 1 to 10 min, means system will wait 1-10 min between 2 calling;
- ✧ Keep “Retry timing” as default, 12sec
- ✧ Set “Video quality” to 0, means lowest required to network;

After all settings have done, touch “Start SIP call testing”, DX monitor will immediately divert a calling to mobile.

2) When calls divert to mobile automatic:

Once pick up the call, check video. (No audio in auto-test), hang up, and close APP, wait for next calling.

3) Check Real-time diagnose statistics:

Once Auto-Test start, DX monitor goes to diagnose statistics ([refers to Part3/Page3](#)), during the test in diagnose page, touch “Unlock” key to check real time diagnose information, touch “Unlock” to get back. Touch “Talk” button to force ending the calling.

4) Copy and Clean diagnose data:

After test check with a general test report in “SIP tools” -> “SIP call statistics”, to back up the diagnose data, goes to “SIP tools” -> “Copy SIP call logs to SD”. Export to excel and save. Clean diagnose by “SIP tools” -> “Clear SIP call statistics” and prepare for next calling test.

5) Test with different network environment

- ✧ Connect DX monitor into different Wi-Fi/Hotpot and repeat step 1) – 4)
- ✧ Use different network on mobile 4G or Wi-Fi and repeat step 1) – 4)

6) Manually test, simulate user scene.

Set DX monitor “Call scene” -> “Divert call always” or “Divert, if no answer”, and call monitor by door station, on mobile pick up call and test with video, video and unlock. Calls back to door station on mobile, to check video, audio and unlock.

## 5. Use “SIP tools” implement automatic testing

DX monitor with firmware version above than “V1.5.0” will provide “SIP tools”, “SIP tools” is simply tool to help customer to test DX monitor SIP divert function automatically and get a clear report. The result will have a clearly view of the quality of SIP divert.

To get in to the “SIP tools” by “Settings” -> “Wireless” -> “SIP tools”. Menu as below:

### 1. SIP calling testing

“SIP calling testing” is test tool that let DX monitor automatic calls to mobile by a certain rules. With settings, the monitor could unattended divert SIP calls to mobile regularly.

Menu show as figure5, available and default value as below:

- ✧ “SIP call testing count” available value 1-100, default: 1
- ✧ “Call interval” available value 1-10, default: 3, suggest no less than 2
- ✧ “Retry timing” available value 12-20, default: 12, please keep as default.

✧ “Video quality” available value 0-3, default: 1. Suggest when testing set as 0

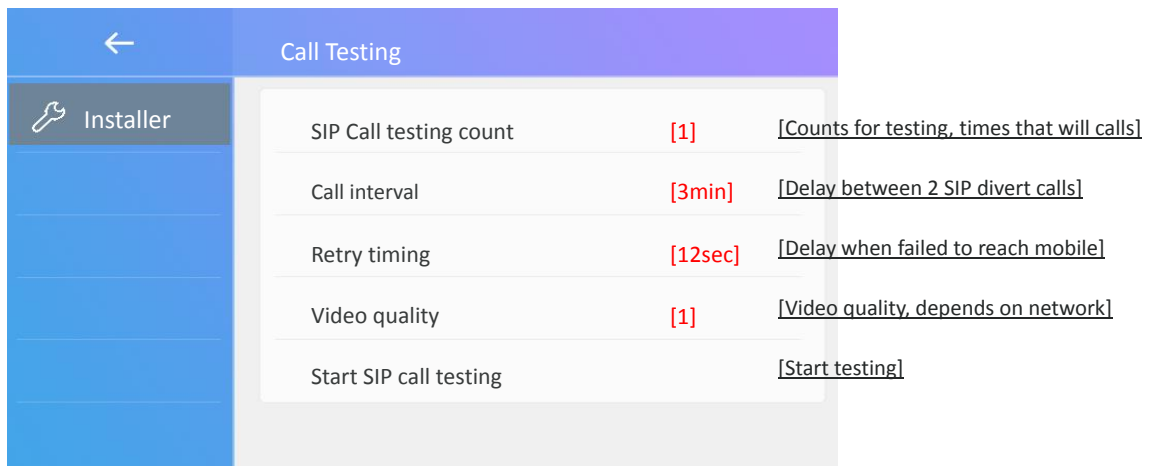


Figure5. Call testing menu

## 2. Call Statistics

Call Statistics shows a general information of all you test before (since last Log clean).

Also if start SIP calls testing, a real time call statistics will shows on screen. ([Refer part3/page3 figure4](#)), in this occasion, only the testing statistics will show, also with tips of the status of testing.

## 3. Clear Sip call statistics

Clear all SIP calls log

## 4. View SIP call logs

Check with all events in the call log, [refer part2/page1](#) to see all text filed means. Loading time will be long depends on call logs.

## 5. Copy SIP call logs to SD

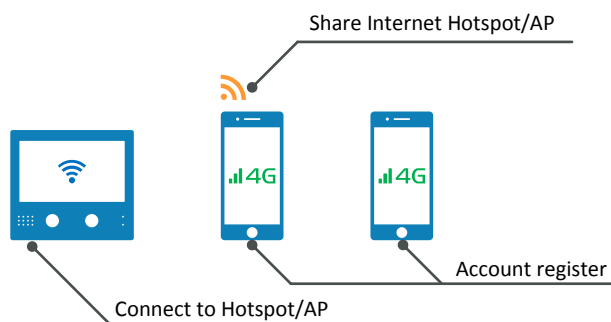
Copy all SIP call logs in text to SD, after you could check it on computer.

## 6. View registered Phones (reserve)

# 6. Test network environments and some tips

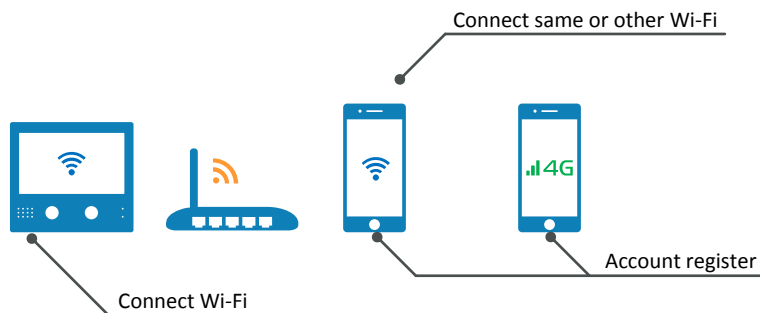
Here is some testing network setup as reference:

1. Quick test with 1 or 2 mobiles and monitor connected with mobiles' hotspot(access point), mobile or 2<sup>nd</sup> mobile login SIP account and test



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2. Test in different network environment, mobile connect with different Wi-Fi or 4G GSM. To test in different network how SIP stability works.



Some useful tips when you testing with DX monitor:

- ✓ When use 2Easy app, disable "IPV6".
- ✓ Use only 2Easy app in testing, otherwise "PUSH" is not available.
- ✓ Delete or "Kill" any other running VOIP APP during test.
- ✓ Setup "Divert Video res" to 0, apply to lower internet bandwidth.
- ✓ After 1 SIP divert, delay 1min to test next.
- ✓ Manual register after DX monitor first connect to Internet.
- ✓ Disable all firewall on "Router", specially "SIP ALG" and "RTSP ALG".

## 7. Export and Analyze SIP Log via "EXCEL"