

Test on IX130 Performance

This document aims to report the performance of IX130. We design a few scenarios where uses IX130, including gateway and PBX and how well the performance of handling transcoding.

Test environments

OpenVox IPC 110 1.6GHz

CPU: Intel® Atom™ CPU Z530 @1.6GHz

Memory: 2GB

OpenVox IPC 110 1.1GHz

CPU: Intel® Atom™ CPU Z510 @1.1GHz

Memory: 2GB

Asterisk-1.8.9.0

WinSIP Version: 2.4.7

TestPC:

CPU: Intel 5300

Memory: 2GB

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Chapter 1 IX130 as a Gateway

Topological Graph

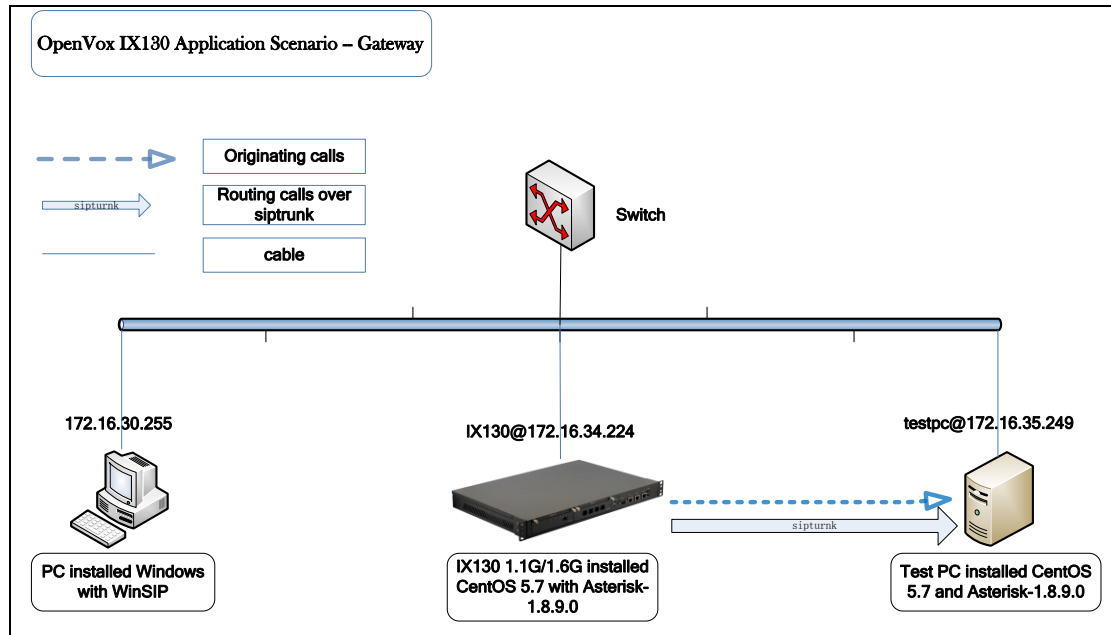


Figure 1 topological graph

Test Steps

1. Install CentOS-5.7 operation system on both machines IX130 and test PC.
2. Download and compile Asterisk-1.8.9.0 in both machines IX130 and test PC.
3. Generate and configure sip.conf and extensions.conf.
4. Set up siptrunk.
5. Configure WinSIP.
6. Originate calls and check related data.
7. Write down test results.

Test Details

We suppose you are done step 1 and step 2. We ignore them here. In order to generate many sip extensions and dialplan information, we come up with a method to generate them in shell script. Of course, you can use other methods to implement as well.

1. Shell script of generating sip extensions information: Bulk_generate_sip.sh

```
#!/bin/sh
#This script is using for generating several sip accounts
#Created on 2012-05-29
for(( a=301; a<=601; a++))
```

```
do
    echo "[$a]"
    echo "type=friend"
    echo "username=$a"
    echo "qualify=yes"
    echo "port=5060"
    echo "nat=yes"
    echo "host=dynamic"
    echo "dtmfmode=rfc2833"
    echo "context=from-internal"
    echo "canreinvite=no"
    echo ""
    echo ""
done
```

2. Shell scrip of generating dialplan. Bult_generate_extensions.sh

```
#!/bin/sh
#This script is using for generating several sip accounts
#Created on 2012-05-29
for(( a=301; a<=601; a++))
do
    echo "exten =>$a,1,Dial(SIP/siptrunk/$a)"
    echo "exten =>$a,2,Hangup()"
    echo ""
    echo ""
done
```

3. Regarding these two scripts, you will also be able to modify them to meet your own requirements.
4. Performing them like this:

```
chmod a+x bulk_generate_sip.sh
./bulk_generate_sip.conf>> sip.conf

chmod a+x bulk_generate_extensions.sh
./bulk_generate_extensions.sh>>extensions.conf
```

After finishing these steps, remember to increase context name for them.

5. Set up siptrunk. Please add these lines to the bottom of sip.conf to set up a siptrunk.

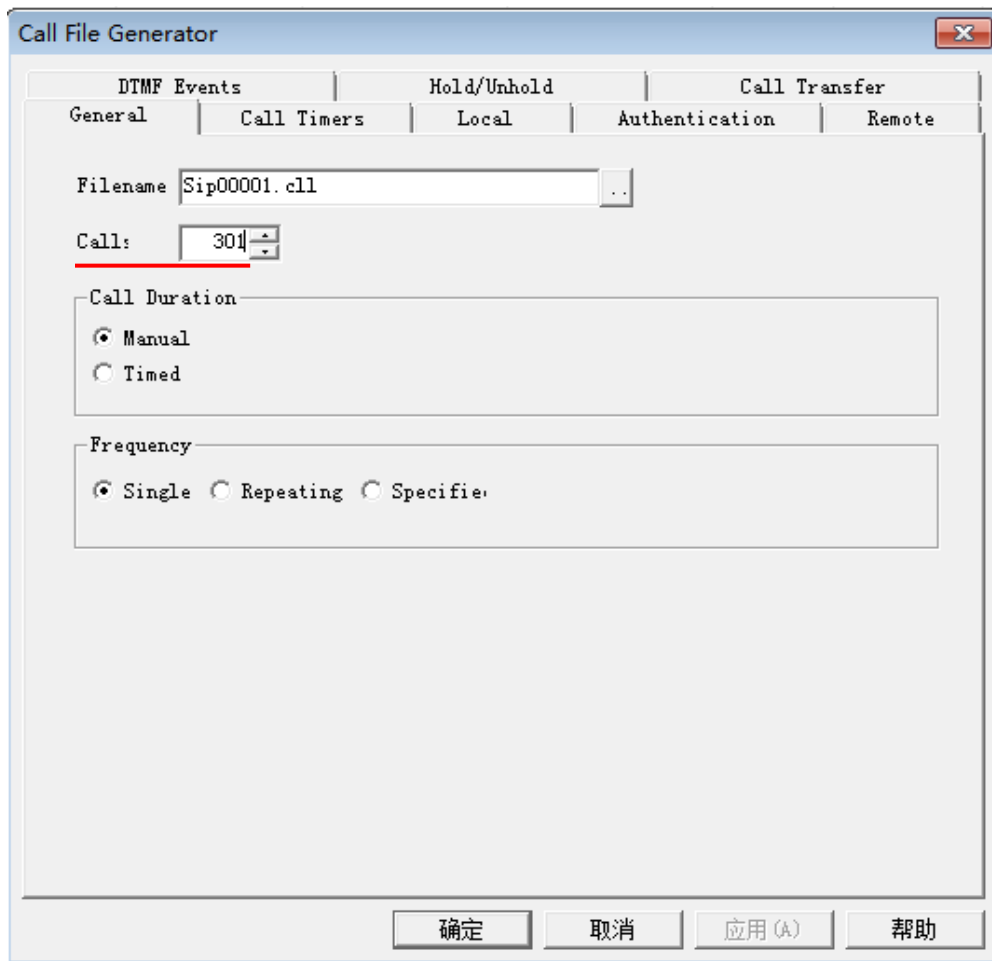
```
[siptrunk]
type=friend
```

```
host=172.16.35.249
context=from-sip
;disallow=all
allow=ulaw
insecure=invite
secret=301
canreinvite=no
fromuser=301
username=301
authname=301
```

Meanwhile, add this line: **register=>301:301@172.16.35.249** to next [general] text, it could be like this:

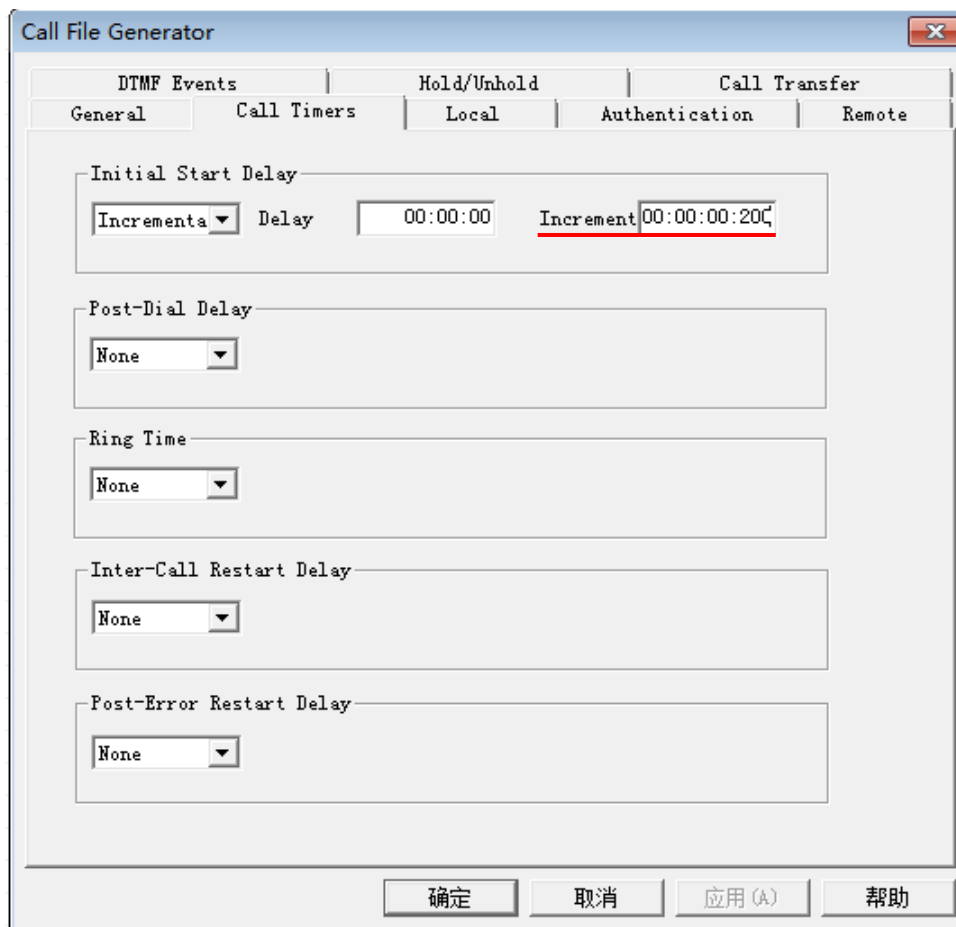
```
[general]
register=>301:301@172.16.35.249
```

6. Configure WinSIP.
 - a. It is supposed that you have already installed WinSIP.
 - b. Once you finished the installation, please start WinSIP and click [File] on the top of the menu tag and its sub-menu [New] to create a new call configuration file. It will pop out a page like that:



As you can see from the [General] tag, what I want to change is [Calls], this option defines how many calls you would like to originate, and here I just take 301 (from extension 301 to 602) as an example. Other options just keep default.

- c. Click [Call Timers] option, a page will be seen as below.



The screenshot shows the 'Call File Generator' dialog box with the 'Call Timers' tab selected. The 'Initial Start Delay' section is expanded, showing a dropdown menu set to 'Increments', a 'Delay' field with '00:00:00', and an 'Increment' field with '00:00:00:200'. The other sections (Post-Dial Delay, Ring Time, Inter-Call Restart Delay, and Post-Error Restart Delay) are collapsed and set to 'None'. The dialog has buttons for '确定' (OK), '取消' (Cancel), '应用(A)' (Apply), and '帮助' (Help).

In the above illustration, what I should change is [Initial Start Delay] option. In my case, I change it to "00:00:00:200", which means that I originate 5 calls in each second.

d. Continue to slide to [Local] option, you will see the following page:

Call File Generator

DTMF Events | Hold/Unhold | Call Transfer

General | Call Timers | Local | Authentication | Remote

ID/Number: 301 Incremental

Address 172.16.30.255 Fixed

Port 5060 Fixed

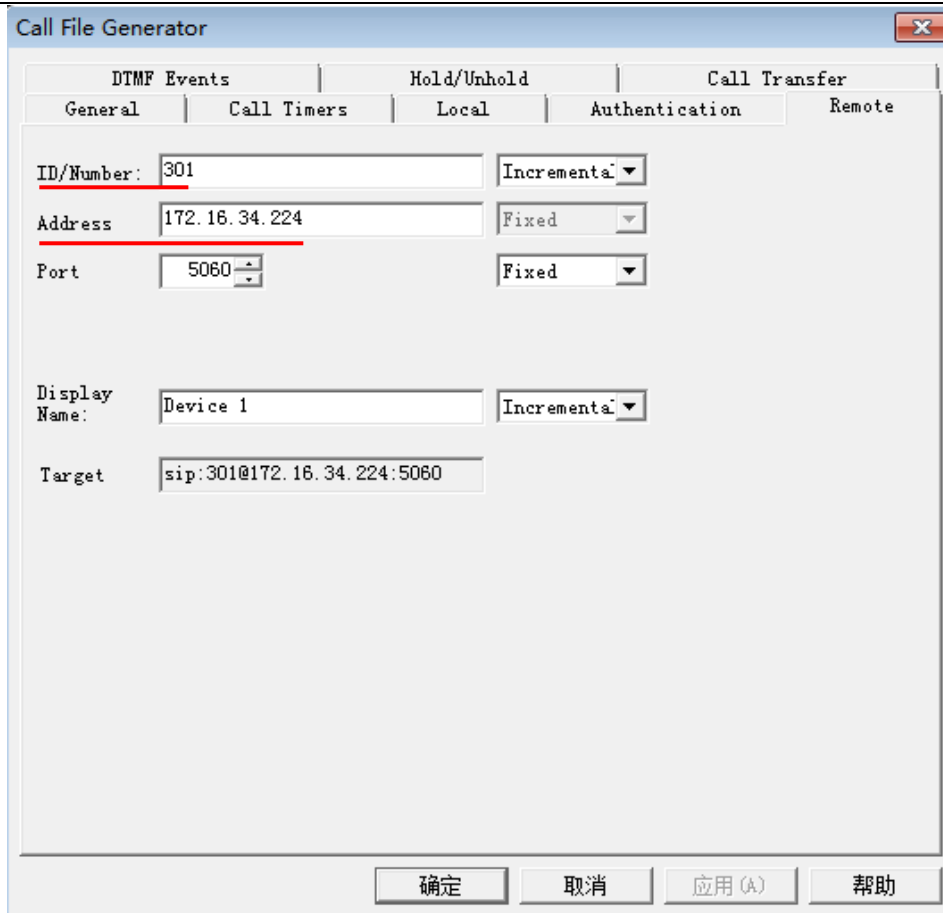
Display Name: WinSIP 1 Incremental

Local URI: sip:301@172.16.30.255:5060

确定 取消 应用(A) 帮助

There are a few options here, in this case, I just need to change [ID/Number] to 301(extension number), and check [Address] option is my local system address.

- e. [Authentication] option, you have to input the username 301<Incremental>, regarding [Password], right here, the password section depends on your sip.conf definition. If you didn't define secret for each sip extension, then [Password] can be ignored here.
- f. [Remote] option, [ID/Number] should be input the starting sip extension 301. [Address] option is a system address that used for originating calls. In this case, it is 172.16.34.244. Other options please keep default.



Call File Generator

DTMF Events | Hold/Unhold | Call Transfer

General | Call Timers | Local | Authentication | Remote

ID/Number: 301 Incremental

Address: 172.16.34.224 Fixed

Port: 5060 Fixed

Display Name: Device 1 Incremental

Target: sip:301@172.16.34.224:5060

确定 取消 应用(A) 帮助

- g. [Call Transfer] option, [ID/Number] should be input 301, and make sure [Address] is 172.16.34.244. Other tags are nothing to do with this test. Click [Apply] and [Yes] to save the configurations.
- h. Click [Options] tag, and scroll it to [Settings]. Find out the [Media] sub-tag, at the Audio Capabilities section; you should select G.711 Ulaw or G.711 Alaw to the [Select] text box. Meanwhile, Outbound Audio Format should be selected G.711 Ulaw 64 kb/s or G.711 or G.711 Alaw 64 kb/s. Let other options be of them.
- i. [Proxy/Register] option, at this page, make sure that [Outbound Proxy] is 172.16.34.244, register is 172.16.34.244 as well. Click [Apply] and [Yes] to confirm the settings. Read [here](#) for more settings about WinSIP.
- j. After setting the relevant options, you will get the sip client ready. Starting both Asterisk system [IX130@172.16.34.244](#) and [testpc@172.16.35.249](#).
- k. On the top of the tag, you will see [Mode] drop-down list, select [Initiate Calls] to originate calls from IX130 to test PC. Then, you can see all the channels' status and how many calls have been started on the panel.

- l. In command line, you can type “top” to check CPU usage ratio of Asterisk. It is a main parameter what performance of the system.

- m. Write down all the related data and input to a data sheet.

Test Results

Result of Concurrent Calls On IPC110					
Version of IPC110	Record Channel Number	Counts of Concurrent Calls	Voice of Quality	Usages of CPU	Types of Hard Disk
1.6 GHz	100	300	Good	197.90%	SATA
1.6 GHz	0	300	Good	159.10%	SATA
1.1 GHz	10	200	Good	168.90%	SATA
1.1 GHz	0	256	Good	199.20%	SATA

Chapter 2 IX130 as a PBX

Topological Graph

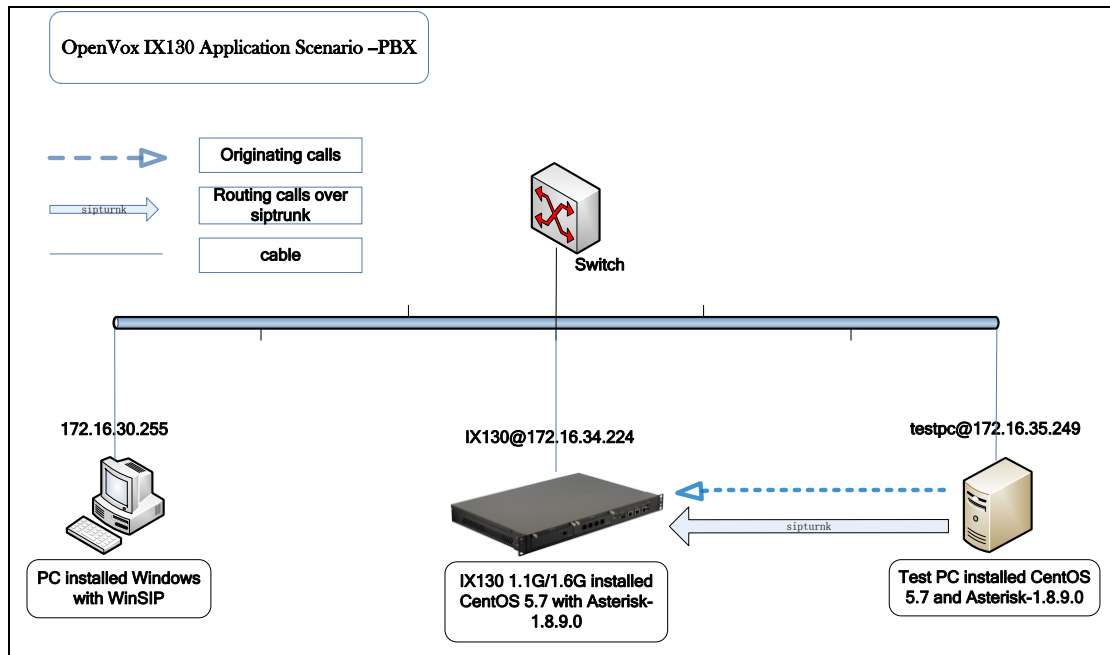


Figure 2 IX130 as a PBX

In order to implement PBX functions, we suppose that IX130 have received calls from test PC. Compare to the previous case, we just change the call routing direction.

Test Results

IX130 as PBX over SIP					
Version of IPC110	Record Channel Number	Counts of Concurrent Calls	Voice of Quality	Usages of CPU	Types of Hard Disk
1.6 GHz	0	300	Good	165.10%	SATA
1.1 GHz	0	256	Good	199.70%	SATA

Chapter 3 IX130 as a Gateway over OpenVox D230

Topological Graph

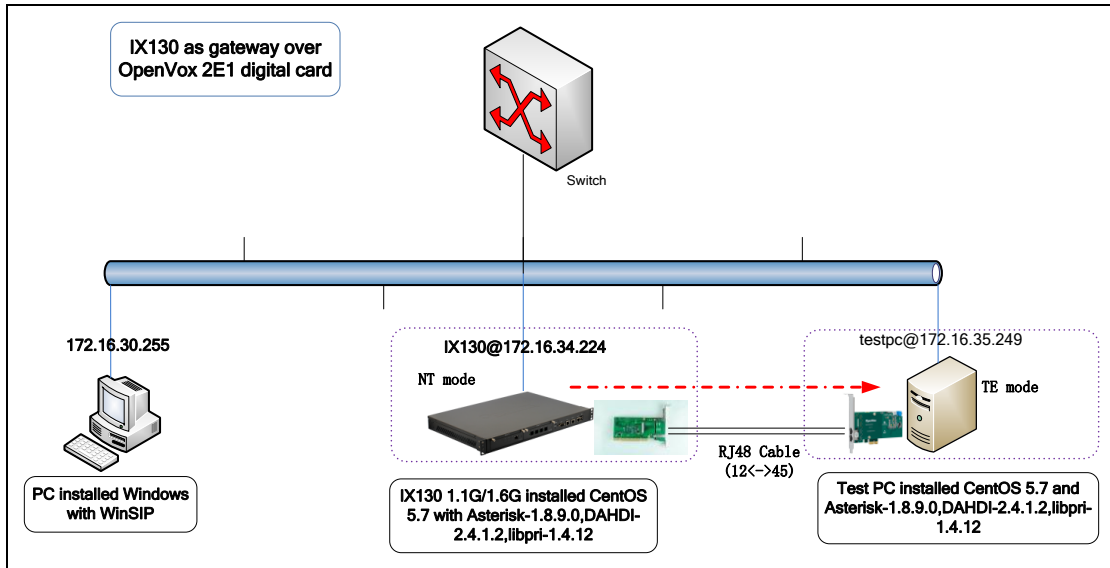


Figure 3 IX130 as gateway over E1 card

In this scenario, we route calls from IX130 to test PC over OpenVox D230. And we configure IX130 as NT terminal, test PC as TE terminal. With the same method above, the test result is as follows.

Test Results

IX130 as Gateway Over 2E1 Digital Card					
Concurrent Calls	Record Channel Number	Version of IX130	Voice of Quality	CPU Usage	Types of Hard Disk
60	0	1.6GHz	Good	55.40%	SATA

Chapter 4 IX130 as PBX Over OpenVox D230

Topological Graph

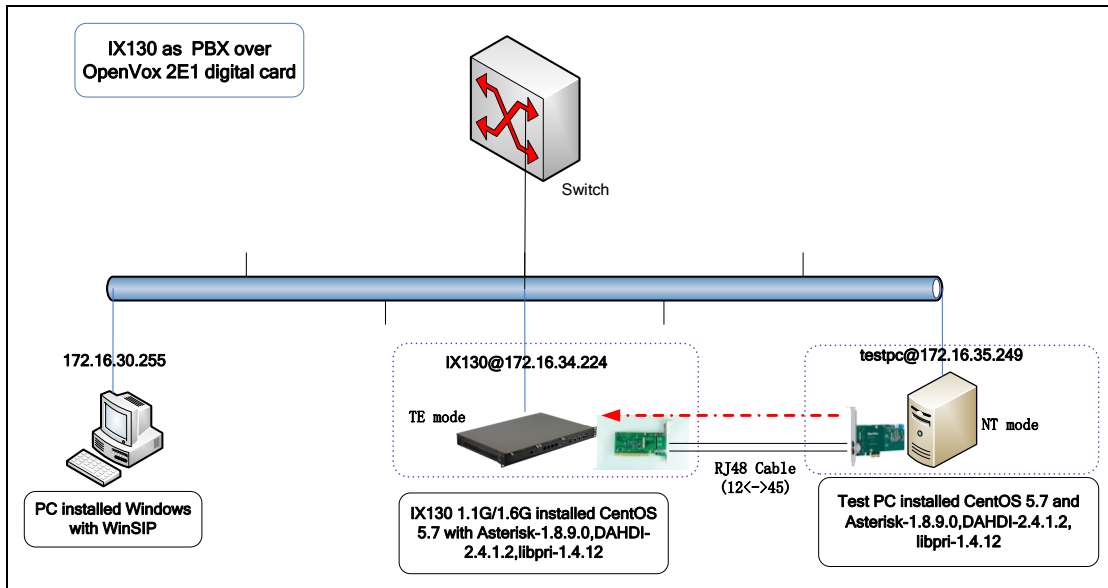


Figure 4 IX130 as PBX over E1 card

In this scenario, we route calls from test PC to IX130 over OpenVox D230. Test PC sets as NT terminal, and IX130 sets as TE terminal.

Test Results

IX130 as PBX Over 2E1 Digital Card					
Concurrent Calls	Record Channel Number	Version of IX130	Voice of Quality	CPU Usage	Types of Hard Disk
60	0	1.6GHz	Good	53.10%	SATA

Chapter 5 Software Transcoding Test with IX130

Topological Graph

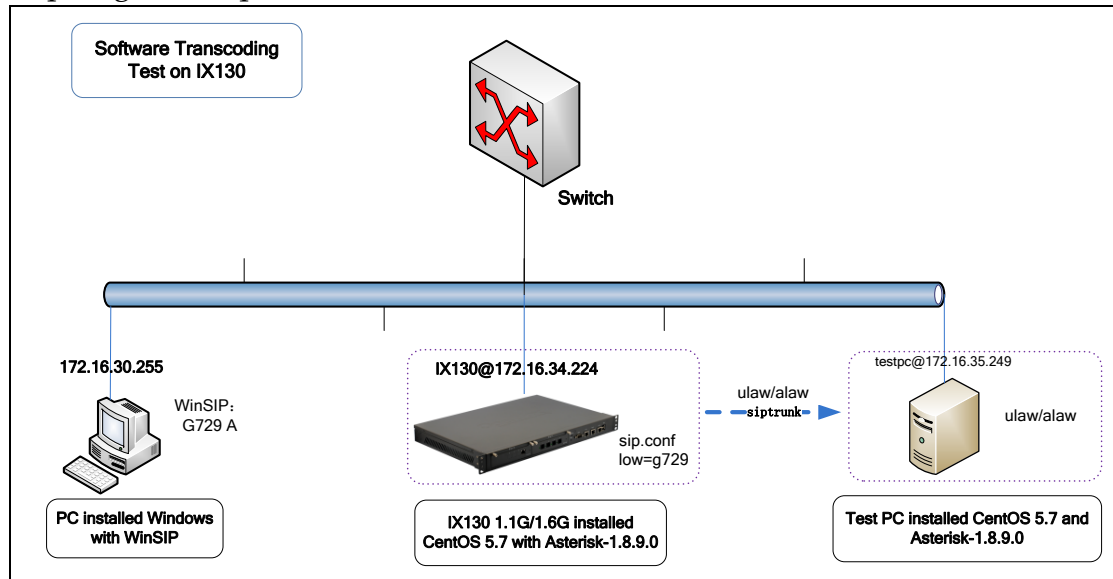


Figure 5 Software Transcoding on IX130

Here are some tests on the highest concurrent calls among g.711 alaw/ulaw, g.722 and g.729. We download codes_g729.so binary from [here](#).

Test Results

Transcoding on IX130			
Type of codes	After Transcoding	Concurrent Calls	Version of IX130
g.711	g.729	65	1.6GHz
g.722	g.729	50	1.6GHz
g.711	g.722	100	1.6GHz

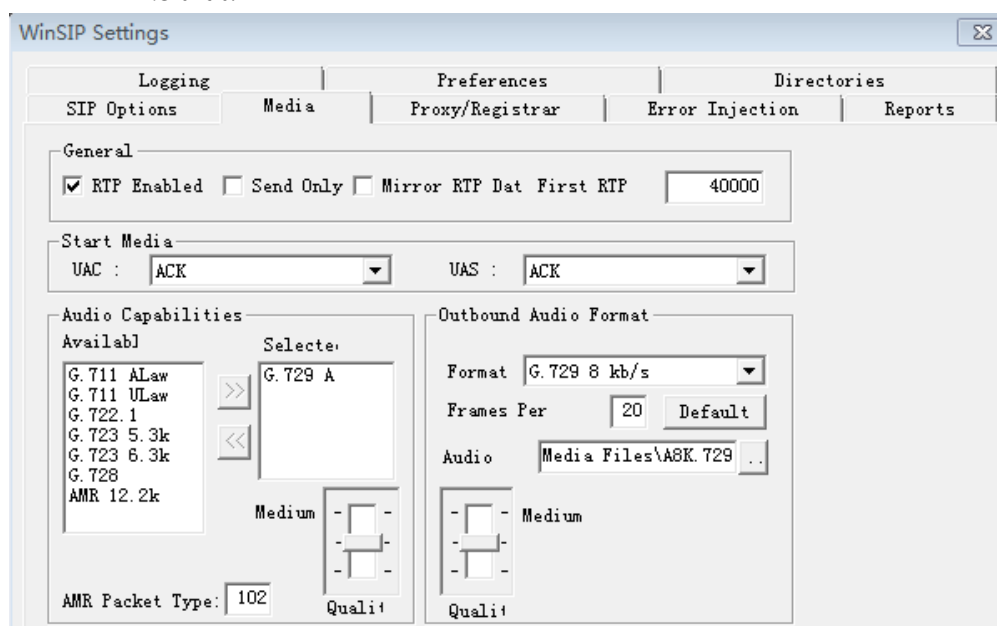
Test Detail Methods

1. With regard to g.729 test, we download g729.so binary from Internet. In our case, IX130 integrates Intel Atom series of CPU, we use CentOS as Operation System, install Asterisk-1.8 on this platform. Therefore, and download [codecs_g729-ast18-gcc4-glibc-atom.so](#) for testing.

2. Change its name from codes_g729-ast-18-gcc4-glibc-atom.so to codes_g729.so, and give executing permission to it by command **chmod a+x codes_g729.so**. Then move it to directory **/usr/lib/asterisk/modules/**.
3. Set configuration files (g.729<-->alaw/ulaw).
 - a. sip.conf - Please add parameter **allow=g729** to each extension definition. For instance, set 301 like that:

```
[301]
type=friend
username=301
qualify=yes
port=5060
nat=yes
host=dynamic
dtmfmode=rfc2833
context=from-internal
canreinvite=no
disallow=all
allow=g729
```

- b. WinSIP configuration-Forward to [Option]->Media->Audio Capabilities. In these two text boxes, please select **G.729 A** for testing. Meanwhile, for “Outbound Audio Format” option, select “G.729 8kb/s”. The finished page will like that:



- c. Keep siptrunk (172.16.34.224 [sip.conf]) settings as their default.

```
[siptrunk]
type=friend
host=172.16.35.249
context=from-sip
disallow=all
allow=ulaw
insecure=invite
secret=301
canreinvite=no
fromuser=301
username=301
authname=301
```

- d. After finishing the above three steps, you will be able to test transcoding between g.711 alaw/ulaw to g.729 in both machines.
- e. Test g.722<-->g.711 by the same method above, what we have to change is the section of each sip extension in sip.conf. We must change parameter "allow" to ulaw. Meanwhile, "allow" value should be changed to g.722 in siptrunk setting of machine 172.16.34.224[sip.conf].
- f. In g.722<-->g.711 test, you need to change is the section of Audio Capabilities. Please change it to g.722 so that the voice is transmitted in g.722 format over siptrunk between these two machines.
- g. Install codes_g.729.so in both machines according to their CPU types in group of g.722<-->g.729 test. In my lab, my test PC's CPU type is Pentium® Dual-Core E5300 @ 2.60 GHz. It is supposed that [codec_g729-ast18-icc-glibc-pentium4.so](#) was the closest type of the CPU.
- h. Regarding to WinSIP settings, change [audio capabilities] to g.729. And siptrunk setting of parameter "allow" should be changed to g.722.
- i. With these steps, you will be able to test three groups of transcoding. Write down all data you want.