



深圳开源通信有限公司

OpenVox V100 User Manual



V100

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Date: 01/08/2011

Version: 1.0





深圳开源通信有限公司

OpenVox-Best Cost Effective Asterisk Cards

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Content

1.	Overview	4
	1.1 What is V100	4
	1.2 What is asterisk	4
2.	Software installation and configuration	5
	2.1 Download	5
	2.2 Installtion	6
	2.3 Call text	9



Test environment

CentOS-5.5 Kernel version: 2.6.18-194.el5 V100: opvx_tc_linux_x86-1.0.0 Asterisk: 1.6.2.11 DAHDI: dahdi-linux-complete-2.4.1.2+2.4.1 Hardware: Openvox V100

1. Overview

1.1 What is V100

V100 is a high density voice transcoding device. Because of low bandwidth requirements, the voice data compression codecs, such as G.729, G.726, iLBC, are commonly used in VoIP applications, the G.711 codecs are widespread in legacy telephone network. The voice signal must be converted in real-time when a call passes through two different networks and each supports its own codec. Compared with transformation in software, V100 makes full use of multicore-DSP, which is able to convert more sessions of different codec modes such as gsm, ilbc, g729, g726, g723, g722, g711. It also reduces bandwidth occupation ratio and relieves system resources.

1.2 What is asterisk

The Definition of Asterisk is described as follows:

Asterisk is a complete PBX in software. It runs on Linux, BSD, Windows (emulated) and provides all of the features you would expect from a PBX and more. Asterisk does voice over IP in four protocols, and can interoperate with almost all standard-based telephony equipments using relatively inexpensive hardware. Asterisk provides Voicemail services with Directory, Call Conferencing, Interactive Voice Response, Call Queuing. It supports three-way calling, caller ID services, ADSI, IAX, SIP, H323 (as both client and gateway), MGCP(call manager only) and SCCP/Skinny(voip-info.org).



V100 user manual



Figure 1 Topology

2. Software installation and configuration

There are three different interface types of V100, which are PCI, PCI-E and RJ45. V100 implements aggregation and distribution mode codec transcoding, so users are able to select a type according to the specific environment. Figures are stated as below.



2.1 Download

1. Download asterisk package from website www.asterisk.org. Right here, take asterisk-1.6.2.11 as an example.

wget http://downloads.asterisk.org/pub/telephony/asterisk/ old-releases/asterisk-1.6.2.11.tar.gz

2. Download DAHDI package by command below

wget http://downloads.openvox.cn/pub/drivers/dahdi-linux-c omplete/openvox_dahdi-linux-complete-current.tar.gz

3. Download V100 package by command below:

wget http://downloads.openvox.cn/pub/drivers/transcoding_c ards/opvx_tc_linux_x86-1.0.0.tar.gz

Release History

Vesrsion	Date	Changes	Editor
1.0	01/08/2011	First preliminary release	Kevin Yi

2.2 Installtion

Please note that if there is no kernel source in the system, users should install it by command "**yum install kernel-devel** ".

Please execute command like "**yum install XXXX**" to check and install dependencies, otherwise system will indicate you that there is nothing to do, please go to the next step.

```
# yum install bison
# yum install bison-devel
# yum install ncurses
# yum install ncurses-devel
# yum install zlib
# yum install zlib-devel
# yum install openssl
# yum install openssl-devel
# yum install gnutls-devel
# yum install gcc // make sure the gcc version is above 4.0
# yum install gcc-c++
```

1. Check the V100 hardware by command: lspci -vvvv

If V100 Ethernet controller is found, you will see outputs as follows:

01:03.0 Ethernet controller: Realtek Semiconductor Co., Ltd. RTL-8139/8139C/8139 Subsystem: Realtek Semiconductor Co., Ltd. RTL-8139/8139C/8139C+ Control: I/O+ Mem+ BusMaster+ SpecCycle- MemWINV- VGASnoop- ParErr- Step Status: Cap+ 66MHz- UDF- FastB2B+ ParErr- DEVSEL=medium >TAbort- <TAbort Latency: 64 (8000ns min, 16000ns max) Interrupt: pin A routed to IRQ 209 Region 0: I/O ports at c800 [size=256] Region 1: Memory at dcdffc00 (32-bit, non-prefetchable) [size=256] Capabilities: [50] Power Management version 2 Flags: PMEClk- DSI- D1+ D2+ AuxCurrent=0mA PME(D0-,D1+,D2+,D3hot Status: D0 PME-Enable- DSel=0 DScale=0 PME-

Figure 4 PCI hardware detection



```
04:00.0 Ethernet controller: Broadcom Corporation NetLink BCM57780 Gigabi Ethernet
PCIe(rev 01)
Subsystem: Broadcom Corporation Unknown device 9692
Control: I/O- Mem+ BusMaster+ SpecCycle- MemWINV- VGASnoop- ParErr- Stepping-
SERR- FastB2B-
Status:Cap+66MHz-UDF-FastB2B-ParErr-DEVSEL=fast>TAbort-<TAbort-<MAbort->SERR-
<PERR-
Latency: 0, Cache Line Size: 32 bytes
Interrupt: pin A routed to IRQ 106
Region 0: Memory at febf0000 (64-bit, non-prefetchable) [size=64K]
```

Figure 5 PCI-E hardware detection

If V100 is not able to be detected, then you need to install the corresponding NIC driver. We offer the V100 NIC driver of PCI-E interface in the directory

../opvx_tc_linux_x86-xxx/eth_drivers/tg3

Once you finish the network driver, perform "**ifconfig**" to check NIC details and ascertain V100 NIC device number according to Hwaddr address, details are showed as following.

```
eth1 Link encap:Ethernet HWaddr A0:98:05:02:00:01  // physical address of V100
inet addr:10.1.1.80 Bcast:10.1.1.255 Mask:255.255.255.0
inet6 addr: fe80::a298:5ff:fe02:1/64 Scope:Link
UP BROADCAST RUNNING MULTICAST MTU:1500 Metric:1
RX packets:0 errors:0 dropped:0 overruns:0 frame:0
TX packets:73 errors:0 dropped:0 overruns:0 carrier:0
collisions:0 txqueuelen:1000
RX bytes:0 (0.0 b) TX bytes:10781 (10.5 KiB)
Interrupt:209 Base address:0xec00
```

Figure 6 V100 NIC information

Attention: Remember to disable SELinux service. Perform "vim /etc/selinux/config", change the value of parameter SELINUX to disabled, and then reboot your computer please.

```
# This file controls the state of SELinux on the system.
# SELINUX= can take one of these three values:
# enforcing - SELinux security policy is enforced.
# permissive - SELinux prints warnings instead of enforcing.
# disabled - SELinux is fully disabled.
<u>SELINUX=disabled</u>
# SELINUXTYPE= type of policy in use. Possible values are:
SELINUXTYPE=targeted
```

Figure 7 SELinux configuration file

2. Installation

1) Install DAHDI

```
# cd /usr/src
# tar -xzvf openvox_dahdi- linux-complete-current.tar.gz
# cd dahdi-linux-complete-XX
# make
# make
# make install
# make config
```

2) Install asterisk



cd /usr/src/
tar -xzvf astersik-XX.tar.gz
cd asterisk-XX
./configure
make
make install
make samples

3) Install V100

```
# cd /usr/src
# tar -xzvf opvx_tc_linux_x86-1.0.0.tar.gz
# cd opvx_tc_linux_x86-1.0.0/libopxtc/
# make install
# cd /usr/src/opvx_tc_linux_x86-1.0.0/codec/asterisk/
# make install
```

4) Modify openvox_codec.conf

vim /etc/asterisk/openvox_codec.conf

A. If using PCI or PCI-e as communication interface, in this situation, please simultaneously plugging V100 and asterisk cards to the same server. Sample configuration file openvox_codec.conf is as follows.

[ethX]	// V100 NIC device name
baseudp=5000	
vocalloaddr=10.1.1.100	// IP of V100 Multicore-DSP chip

If your V100 is recognized as eth1 by the system, then you need to change the ethX to eth1. One thing you need to note is place V100 NIC IP and vocalloaddr IP to same network segment, and make these two IPs are different.

B. If using RJ45 as communication interface, please directly ignore step 5 below. In this case, you will need another PC to provide power to V100. Plug V100 to the PC, and then connect the asterisk server with the PC over cable or other network devices. Sample of configuration file openvox_codec.conf are as follows.

[ethX] baseudp=5000 Vocalloaddr=192.168.2.186 // Make sure the IP is available

The X in ethX means the network device that connects with asterisk server. For example, if your server has two network interface cards, one is eth0, and the other is eth1, and suppose to connect eth1 with V100, then you will need to modify vocalloaddr to the same network segment as eth1.

5) Setting V100 NIC IP by running the below command. Right here, take eth1 as an example.

vim /etc/sysconfig/network-scripts/ifcfg-eth1

set **BOOTPROTO =static** add two line: **IPADDR=10.1.1.80**

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NETMASK=255.255.255.0

Realtek Semiconductor Co., Ltd. RTL-8139/8139C/8139C+
DEVICE=eth1
BOOTPROTO=static
ONBOOT=yes
HWADDR=a0:98:05:02:00:01
NETMASK=255.255.255.0
IPADDR=10.1.1.80

Figure 8 ifcfg-eth1 configurations

After the network settings, remember to perform "**service network restart**" to restart the network and activate V100 NIC. See more V100 details by "**ifconfig**".

6) Before starting asterisk, please run "vim /etc/asterisk/modules.conf", and add a line "noload => res_timing_pthread.so" at the end of modules.conf, it will disable the timing module. Otherwise, it's going to display many errors from asterisk.

7) Enable asterisk by running "**asterisk** –**vvvvvvgc**", if it has been started before, run "**asterisk** -**r**" instead. In the CLI, perform "**module load codec_openvox.so**" to load V100 driver.

After entering into CLI, type "**op**" and press Tab. If it displays openvox, which means installation finished elementarily. Please also perform other commands to check related information, for instance, run "**openvox show translators**" to show supportive code conversion mode.

```
*CLI> openvox show translators

ilbc to g726

g726 to ilbc

g723 to g726

.

.

ulaw to g722

g729 to ulaw

ulaw to g729
```

It will show license information as below after run "openvox show license".

*CLI> openvox show license License info: max=256, current=0.

2.3 Call text

Run command below to register two SIP phones, and add configuration at the end of sip.conf.

vim /etc/asterisk/sip.conf



allow=all canreinvite=no

[666] type =friend user=666 secret=666 host=dynamic context=from-internal allow=all canreinvite=no [888] type=friend user=888 secret=888 host=dynamic context=from-internal

Figure 9 SIP phone register

Add dial plan at the end of extensions.conf.

vim /etc/asterisk/extensions.conf

```
[from-internal]
exten=>666,1,Dial(sip/666)
exten=>666,2,Hangup()
exten=>888,1,Dial(sip/888)
exten=>888,2,Hangup()
```

Figure 10 dial plan

Follow the dial plan above to configure two SIP phone, one choose G711 alaw/ulaw as audio encoding pattern, and the other choose G729. If call normally, it means installation is successful.



3. Reference

www.openvox.cn

www.digium.com

www.asterisk.org

www.voip-info.org

www.asteriskguru.com

Tips

Any questions during installation, please consult to BBS or login wiki for answers. Websites are as follows.

http://bbs.openvox.cn/

http://wiki.openvox.cn/index.php/%E9%A6%96%E9%A1%B5