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Realtek VoIP Solution
Menuconfig
RTL89xxC & RTL89xxD serials

Project Headline

RTL89xxC is Realtek SoC with 5 ports Gigabit Ethernet switch solution, while RTL89xxC is Realtek SoC with 5 ports Fast Ethernet switch solution. This project is based on RTL89xxC and RTL89xxD adding DSP command sets and Realtek Software DSP solutions. This project aims cost sensitive market and provides a high efficient VoIP solution.

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USING THIS DOCUMENT

This document provides detailed user guidelines to achieve the best performance when implementing the Realtek Home Gateway Controllers.

Though every effort has been made to ensure that this document is current and accurate, more information may have become available subsequent to the production of this guide. In that event, please contact your Realtek representative for additional information that may help in the development process.

Reviewers

Department	Name/Title
System Design Dept IV	Thlin

Modification History

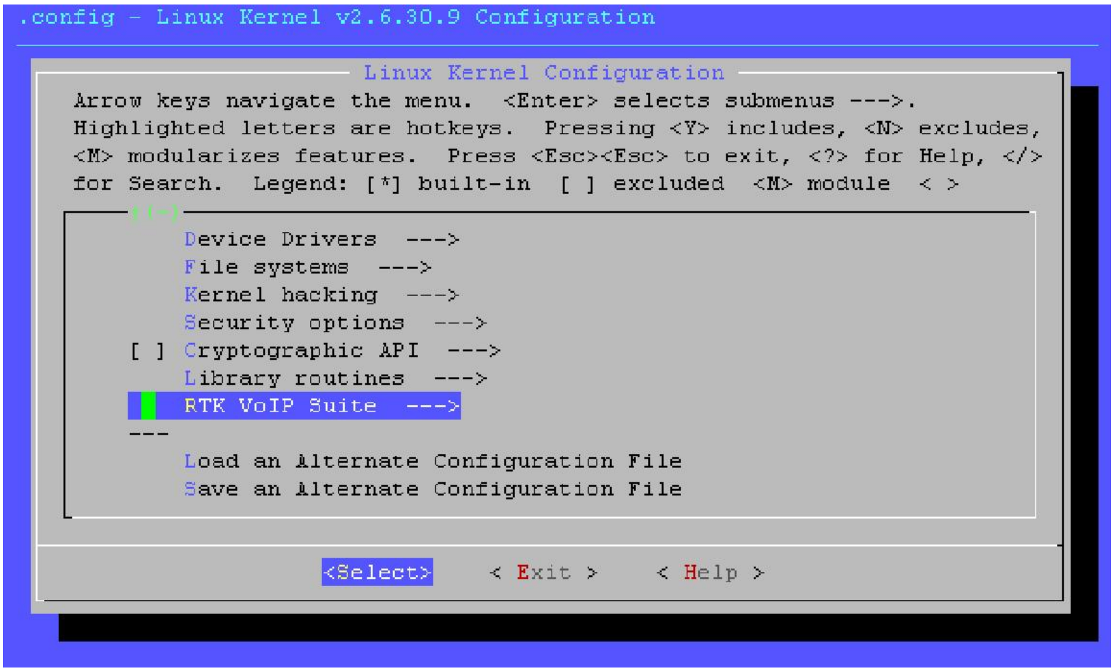
Rev.	Date	Originator	Comment
0.1	10/30/2013	Thlin	Initial revision

1.1 VoIP configuration

Below section describe VoIP relative configurations in RTK VoIP suite.

1.1.1 Enter VoIP's options

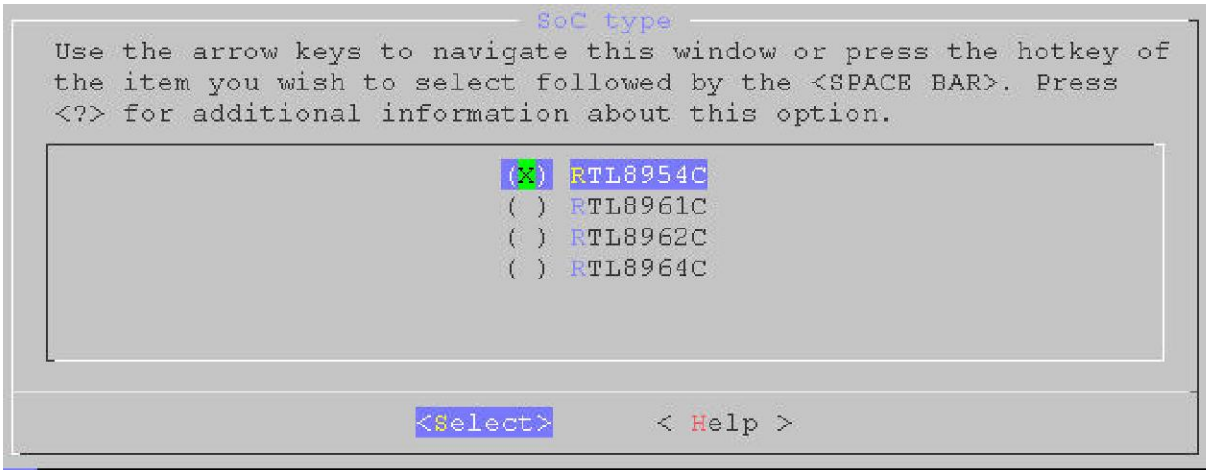
Press 'make menuconfig', and choose 'RTK VoIP Suite" to customize VoIP's options.



1.1.2 SoC type

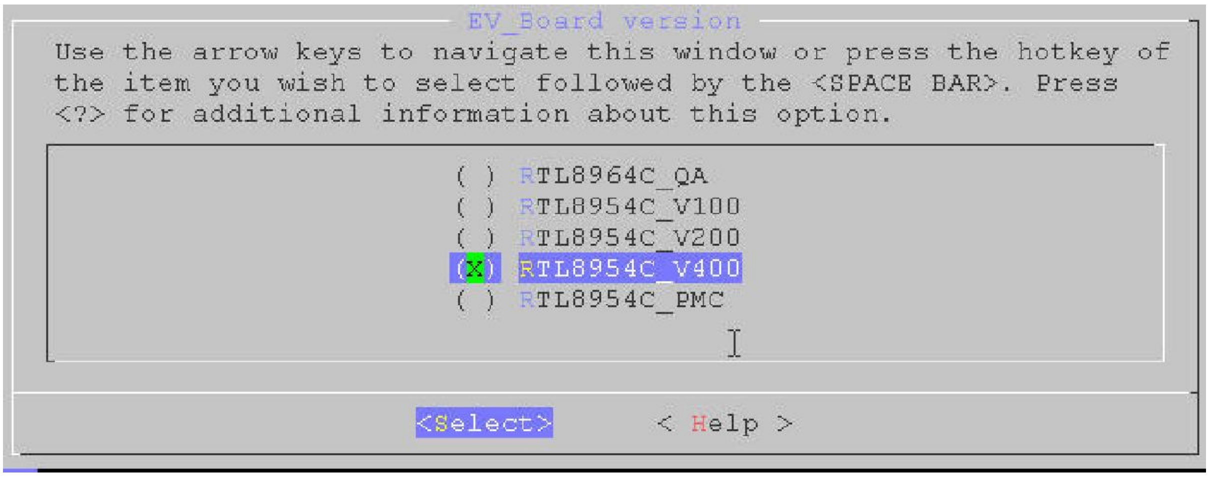
8954C: Use Realtek's demo DSP

8961C/8962C/8964C: Use Audiocodes' official DSP, and support 1/2/4 ports respectively.



1.1.3 EV_Board version

Which EV board do you use?
This option defines GPIO layout. V400 includes all 4xx series.



1.1.4 RTK VoIP support

[*] RTK_VOIP Support

RTK_VOIP allow user to remove all VoIP modules in Linux kernel. Don’t forget to remove “rtk_voip” in user applications when you remove VoIP support from kernel.

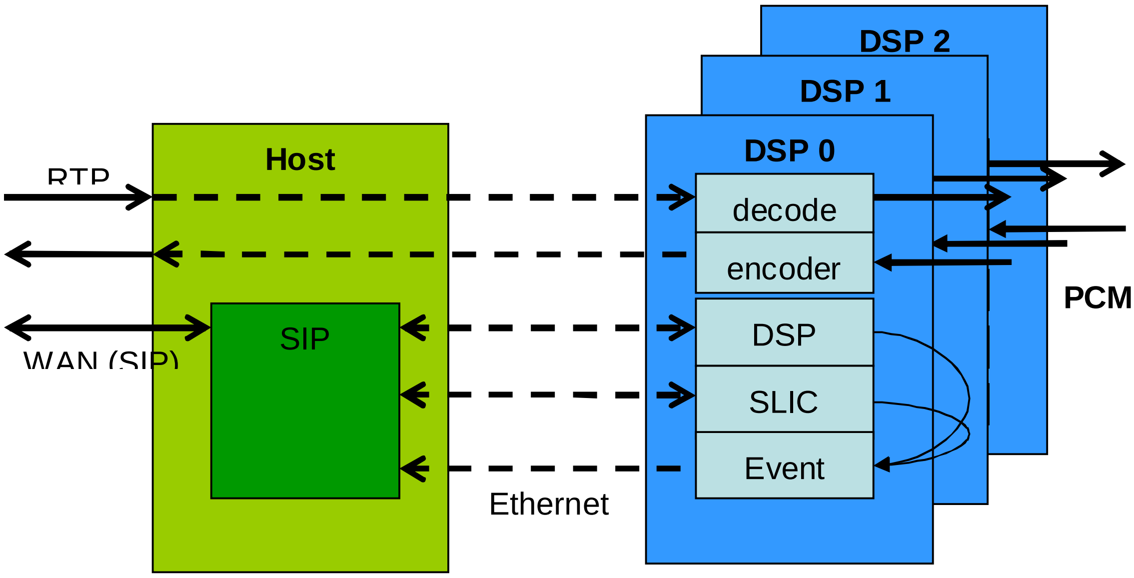
1.1.5 DSP Architecture

[*] Ethernet DSP
Ethernet DSP Type (HOST) --->

Realtek VoIP SDK allow user to connect more then one DSP chips via Ethernet interface. Realtek SDK provides two kinds of DSP architectures. They are standalone, Ethernet DSP. Normally, standalone architecture was used.

1.1.5.1 Ethernet DSP

Ethernet DSP is IPC (distribution) architectures. IPC type contains exactly one host CPU and more then one DSP processors. Ethernet DSP architecture allows customer to design a high density VoIP board.



1.1.5.2 IPC architecture

If customer decides to use IPC architecture, he should build two kinds of firmware "Host" and "DSP" side individually.

```
[*]        Ethernet DSP
               Ethernet DSP Type (HOST)    --->
```

1.1.5.3 IPC architecture – Host side

NFBI Host Driver: This is non-flash booting driver which designs to boot up DSP from host side. (Ethernet DSP only)

Number of DSP Device: How many DSPs are connected.

SLIC/DAA Channel Number Per DSP Device: How many SLIC/DAA are controlled per DSP. It assumes that SLIC/DAA numbers are the same in all DSP, so the amount of channel number is

(DSP Device) * (SLIC per DSP + DAA per DSP)

```
[*]        Ethernet DSP
               Ethernet DSP Type (HOST)    --->
[*]        NFBI Host Driver
(4)         Number of DSP Device
(8)         SLIC Channel Number Per DSP Device
(0)         DAA Channel Number Per DSP Device
```

1.1.5.4 IPC architecture – DSP side

No additional options for DSP side. It is almost same as standalone architecture.

```
[*]        Ethernet DSP
Ethernet DSP Type (DSP)    --->
```

1.1.6 Support FXS/FXO LED

Use Realtek SoC GPIO to controtl FXS/FXO LED. Zarlink SLIC has 4 programmable I/O pin per chip. If you want to use SLIC I/O pin, please disable this one. We will introduce it later.

```
[*]     Support FXS/FXO LED (GPIO)
```

1.1.7 Sound category

Realtek SDK classifies sound devices into three categories:

- **ATA/IAD/SLIC**: SLIC and DAA drivers provided by SiLabs and Zarlink.
- **IP phone**: ALC5621 provide by Realtek
- **ATA/DECT**: DECT drivers provided by SiTEL and DSPG.

```
        *** Sound category for ATA/IAD ***
[ ]     ATA/IAD/SLIC
        *** Sound category for IP Phone ***
[ ]     IP phone
        *** Sound category for ATA/DECT ***
[ ]     ATA/DECT
```

1.1.7.1 Sound category – ATA/IAD

ATA/IAD is a typical VoIP device to bridge traditional analog phone to IP network. SLIC (Subscriber Line Interface Circuit) is the key interface to an analog telephone. Realtek VoIP SDK provides Silab and Zarlink SLIC for reference. So, in this category, you will see 'Silab', 'Zarlink' and 'VIRTUAL DAA'.

```
[*]     ATA/IAD/SLIC
[ ]     Silab
[ ]     Zarlink
[ ]     VIRTUAL DAA
```

1.1.7.2 Sound category – ATA/IAD - Silab

Realtek SDK provides SI3217x (BB and FB[1]), SI3226 (BB) and SI3226x (FB) drivers.

One thing you have to keep in mind, Realtek SDK assume that Silab SLIC use SPI daisy-chain to connect multiple SLICs.

Daisy chain mode allows communication with banks of up to 8-32 ProSLIC

[1] BB and FB are short for "Buck Boost" and "Fly Back" respectively.

devices using one chip select signal. All SLICs in the same chain are sharing one CS pin. The only exception is DAA (FXO). DAA doesn't have capability to connect another SLIC. In other words, DAA must be the last device in the chain. In Realtek VoIP SDK SI32178 is the only device has DAA port.

```
[*]     ATA/IAD/SLIC
[*]     Silab
[ ]       SI3217x
[ ]       SI3226(BB)
[ ]       SI3226x(FB)
```

SI3217x (Buck Boost**):**

Configure SI3217x DC-to-DC converter as "Buck Boost" or "Fly Back". All Silab SI3217x devices assume to use the same DC-to-DC converter.

```
(X) Buck Boost
( ) Fly Back
```

SI32176 number:

Number of "SI32176" connected to SPI daisy-chain. SI32176 is a SLIC with single FXS port.

SI32176 use PIN_CS#:

Use which GPIO pin as CS. In SPI daisy-chain mode, it needs merely one CS.

```
[*]          SI32176 (Daisy Chain)
(1)            SI32176 number
(1)            SI32176 use PIN_CS#
```

SI32178 number:

Number of "SI32178" connected in SPI daisy-chain mode. SI32178 is a SLIC with 1FXS + 1FXO port.

SI32178 use PIN_CS#:

Because SI32178 has a DAA which is a SPI daisy-chain termination, multiple SI32178 must have multiple chip select. It requires more then one GPIO CS pins starting from this configuration.

For example, four SI32178 and PIN_CS# starting from PIN_CS1, and then PIN_CS2 / / CS3 / CS4 are used to control SI32178.

For CS1/CS2/CS3/CS4 definition, please refer to linux-2.6.30/rtk_voip/voip_drivers/gpio/ gpio_8954c_vxxx.h.

```
[*]          SI32178 (Chip Select)
(2)             SI32178 number
(1)             SI32178 use PIN_CS#
```

32176 number among multiple SI32176 + single SI32178:

This is a special option to connect multiple SI32176 and SI32178 in one SPI daisy-chain mode. In this case, SI32178 can be one and the only one. The physical position of SI32178 must be at the end of SPI daisy-chain. Thus, we only need to decide the number of SI32176.

```
[*]          multiple SI32176 + single SI32178 (Daisy Chain)
(3)             32176 number among multiple SI32176 + single SI32178
(1)             multiple SI32176 + single SI32178 use PIN_CS#
```

Above example defined 3 SI32176 + 1 SI32178. So, we totally have 4 FXS + 1 FXO port.

SI3226 number:

Number of “SI3226” connected in SPI daisy-chain mode. SI32176 is a SLIC with dual FXS port. We assume SI3226 use Buck-Boost DC-to-DC converter. It follows Silab reference design.

SI3226 use PIN_CS#:

Use which GPIO pin as CS. In SPI daisy-chain mode, it needs merely one CS.

```
[*]        SI3226(BB)
(1)           SI3226 number
(1)           SI3226 use PIN_CS#
```

1.1.7.3 Sound category – ATA/IAD - Zarlink

Realtek VoIP SDK supports several Zarlink SLIC. There are LE88221/ LE88266/ LE88286, LE88111, LE89116 and LE89316. Zarlink’s options are simple and all chips’ options are similar. Each SLIC occupies one CS pin.

```
[*]      Zarlink
[ ]         LE88221/88266/88286
[ ]         LE88111
[ ]         LE89116
[ ]         LE89316
```

LE882xx number:

Number of LE882xx is mounted. LE88221/LE88266/LE88286 are dual FXS

SLIC. LE88221 and LE88266 are pin to pin compatible.

LE882xx use PIN_CS#:

LE882xx's CS pin starts from this number.

LE882xx use IO1/IO2 as its LED0/Relay:

Zarlink SLIC provides 4 programmable I/O pins per chip. This options will use Zarlink I/O pin to control LED and Relay if has. This feature doesn't work on Realtek EV board due to PCB layout.

```
[*]     LE88221/88266/88286
(2)       LE882xx number
(1)       LE882xx use PIN_CS#
[ ]       LE882xx use IO1/IO2 as its LED0/Relay
```

Above setting defines 2 LE882xx SLIC. It implies we will have 4 FXS ports under this configuration. It also says PIN_CS# starts from CS1. It imply that Realtek SoC will use CS1 to control the first LE882xx and use CS2 to control second LE882xx.

For CS1/CS2/CS3/CS4 definition, please refer to linux-2.6.30/rtk_voip/voip_drivers/gpio/ gpio_8954c_vxxx.h.

LE88111 number:

Number of LE88111 is mounted. LE88111 is a single FXS SLIC.

LE88111 use PIN_CS#:

LE88111's CS pin starts from this number. For CS1/CS2/CS3/CS4 definition, please refer to linux-2.6.30/rtk_voip/voip_drivers/gpio/ gpio_8954c_vxxx.h.

LE88111 use IO1/IO2 as its LED0/Relay:

Zarlink SLIC provides 4 programmable I/O pins per chip. This options will use Zarlink I/O pin to control LED and Relay if has. This feature doesn't work on Realtek EV board due to PCB layout.

```
[*]     LE88111
(1)       LE88111 number
(3)       LE88111 use PIN_CS#
[ ]       LE88111 use IO1/IO2 as its LED0/Relay
```

LE89116 number:

Number of LE89116 is mounted. LE89116 1 a single FXS device.

LE89116 use PIN_CS#:

LE89116's CS pin starts from this number.

LE89116 use IO1/IO2 as its LED0/Relay:

Zarlink SLIC provides 4 programmable I/O pins per chip. This options will use Zarlink I/O pin to control LED and Relay if has. This feature doesn't work on Realtek EV board due to PCB layout.

```
[*]      LE89116
(1)        LE89116 number
(3)        LE89116 use PIN_CS#
[ ]        LE89116 use IO1/IO2 as its LED0/Relay
```

For CS1/CS2/CS3/CS4 definition, please refer to linux-2.6.30/rtk_voip/voip_drivers/gpio/ gpio_8954c_vxxx.h.

LE89316 number:

Number of LE89316 is mounted. LE89316 has 1 FXS + 1 FXO device.

LE89316 use PIN_CS#:

LE89316's CS pin starts from this number.

LE89316 use IO1/IO2 as its LED0/Relay:

Zarlink SLIC provides 4 programmable I/O pins per chip. This options will use Zarlink I/O pin to control LED and Relay if has. This feature doesn't work on Realtek EV board due to PCB layout.

```
[*]      LE89316
(1)        LE89316 number
(4)        LE89316 use PIN_CS#
[ ]        LE89316 use IO1/IO2 as its LED0/Relay
```

For CS1/CS2/CS3/CS4 definition, please refer to linux-2.6.30/rtk_voip/voip_drivers/gpio/ gpio_8954c_vxxx.h.

1.1.7.4 Sound category – ATA/IAD – Virtual DAA

Use a simple hardware circuit to simulate DAA, and its limited functions are followings:

- Use relay to connect FXS and FXO, and SLIC is disconnected.

- Use a circuit to detect FXO ring

```
[*]    VIRTUAL DAA
[ ]      SUPPORT 2 RELAY (NEW)
```

1.1.8 Sound category – IP Phone

ALC5621 is provided for IP phone application. It uses IIS precedence over PCM, and wideband option only exists when IIS support.

Realtek ALC5621 support wideband: Configure as wideband mode.

```
*** Sound category for IP Phone ***
[*]    IP phone
         Audio CODEC (Realtek ALC5621)  --->
```

1.1.9 Sound category – ATA/DECT

Realtek VoIP SDK provides SiTEL and DSPG drivers for DECT application. SiTEL and DSPG are controlled by SPI and UART interfaces respectively.

Interface: UART or SPI

DECT Module: DSPG CMBS Module or SiTEL CVM480 Module

DECT UART Baudrate: UART baudrate for DSPG

DECT UART Flow Control Support: Enable UART hardware flow control

SiTEL handset number: Number of handsets are supported by SiTEL?

```
[*]    ATA/DECT
Interface (UART)  --->
DECT Module (DSPG CMBS Module)  --->
(115200) DECT UART Baudrate
[ ]    DECT UART Flow Control Support
```

1.1.10 Bus configuration

We support PCM and IIS bus for TDM data. Most devices only support PCM bus for data transfer, but ALC5261 supports both PCM and IIS interface.

PCM bus channel number: PCM controller has 32 timeslots for TDM. But Maximum 8 PCM channels work at the same time. This is the limitation of 8954C PCM bus. Enter appropriate number to save memory usage.

In narrow band mode, PCM bus channel number is the same as FXS + FXO port. But in wideband mode, you have to double PCM bus channel number.

```
        *** Bus configuration ***
[*]     Support PCM bus
(8)         PCM bus channel number
[*]     Support IIS bus
```

1.1.11 Application options

Wideband application:
Enable Wideband support. ATA always use narrowband as default codec. SLIC and SDP switch to wideband when wideband codec is selected, for example G.722.

Not all sound devices support ‘wideband’ mode, so device has its own way to explain this option. In other words, ATA allows narrowband and wideband sound device work simultaneously.

Note: You MUST double PCM channel number in wideband mode.

Control channel number: EV board’s FXS + FXO port number.

```
[*]     Wideband application
(3)      Control channel number
```

1.1.12 DSP options

DSP package: Realtek VoIP SDK provides Realtek demo DSP and Audiocodes official DSP. Choose 8961C/8962C/8964C SoC to use Audiocodes’.

```
      *** DSP options ***
DSP package (Realtek)   --->
```

1.1.13 DSP package options – codec

G.711 always enabled, so there is no option for G.711.

G.729AB: Enable G.729

G.723.1: Enable G.723.1

AMR-NB: Enable AMR narrowband

SPEEX-NB: Enable SPEEX narrowband

G.726: Enable G.726

GSM-FR: Enable GSM-FR

iLBC: Enable iLBC

G.722: Enable G.722

T.38 (FAX): Enable T.38

```
          *** Speech CODECs ***
[*]       G.729AB
[*]       G.723.1
[ ]       AMR-NB
[ ]       SPEEX-NB
[*]       G.726
[*]       GSM-FR
[*]       iLBC
[*]       G.722
[*]       T.38 (FAX)
```

1.1.14 DSP package options – others

Enable RTCP XR (RFC3611): Enable RTCP-XR

Enable SRTP: Enable SRTP

Enable SIP TLS: Enable SIP TLS

```
          *** Protocol and Security ***
[*]       Enable RTCP XR (RFC3611)
[ ]       Enable SRTP
[ ]       Enable SIP TLS
```

1.1.15 Networking options

Port Link Monitor: When Ethernet port is connected or disconnected, signal netlink event to 'solar'. This event triggers SIP register or reregister.

Gigabyte Phy Link Mode (10/100/1000): Use gigabyte mode or 10/100 mode

Enable QoS for VoIP: Enable QoS for VoIP, so networking traffic will not affect voice quality.

```
          *** Network ***
[*]       Port Link Monitor
[*]       Enable QoS for VoIP
```