VoIP Setup

Introduction

Voice over IP network (VoIP in short) is the trend to transition of PSTN toll quality voice services to IP network.

There are many different call signal protocols existing. The most popular protocols are SIP, H.323, MGCP and megaco. All these protocols are not compatible to each other unless via soft-switch server. That's why the 2600V series will support all of them by stages.

The first stage of Vigor2600V series support SIP protocol only. This protocol is a replacement protocol of the aging H.323 protocol. It can support peer-to-peer direct calling and also can support calling via sip proxy server (a role similar to gatekeeper in H.323 network). The second stages will implement MGCP protocol. It is a slave-master architecture protocol. The calling scenario is very similar to current PSTN network. The H.323 and megaco protocols will be added after MGCP complete.

After a call is setup, the voice stream transmit via RTP (Real-Time Transport Protocol). Different CODEC can be embedded into RTP packets. We provide G.711 A-law, G.711 mu-law, G.729 A & B, G.723.1 and G.726. Each CODEC consume diferent bandwidth and provide different voice quality. The more bandwidth it consume the better voice quality it produce.However, it still depends on your ADSL stream speed, especially the up stream speed.

This chapter explains the capabilities of VoIP on the router. Use the following setup links on the Setup Main Menu to setup VoIP functions.

Advanced Setup >> VoIP Settings

After you click VoIP Settings link, you will enter the page like Fig.1 :

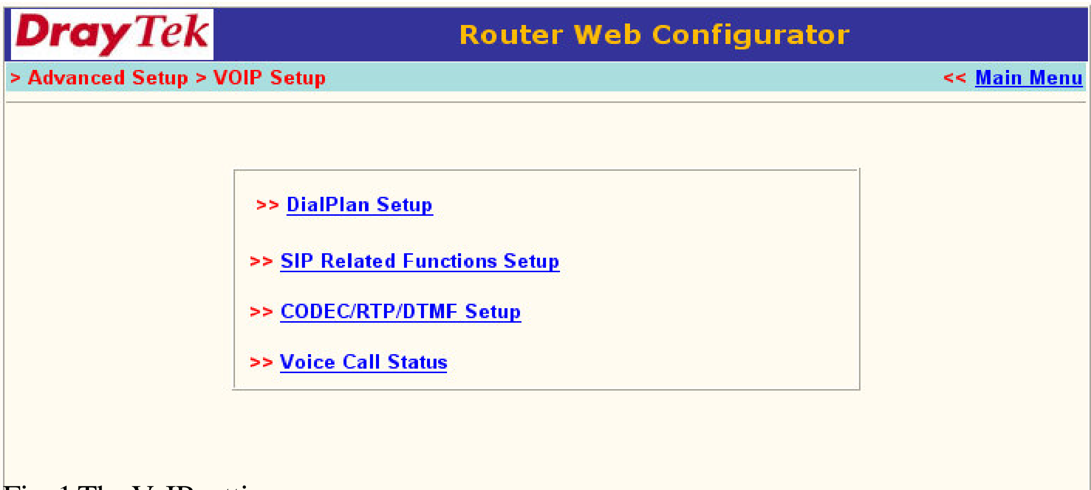
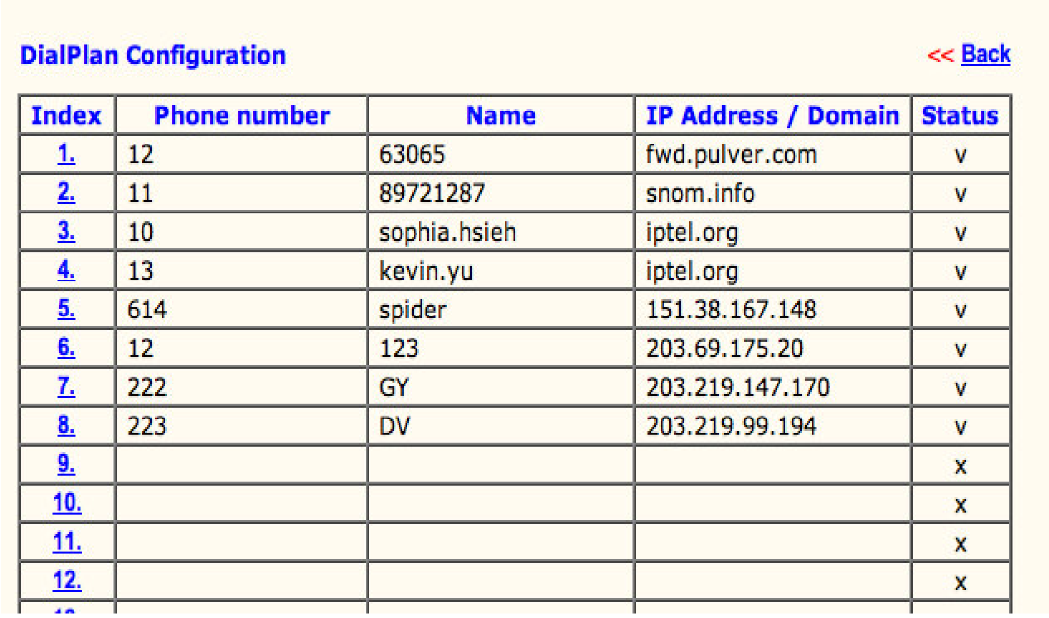


Fig. 1 The VoIP setting page

DialPlan Setup

Click the Dial-Plan Setup, you can setup speed dial phone book up to sixty entries.



DialPlan Configuration << Back

Index	Phone number	Name	IP Address / Domain	Status
1.	12	63065	fwd.pulver.com	v
2.	11	89721287	snom.info	v
3.	10	sophia.hsieh	iptel.org	v
4.	13	kevin.yu	iptel.org	v
5.	614	spider	151.38.167.148	v
6.	12	123	203.69.175.20	v
7.	222	GY	203.219.147.170	v
8.	223	DV	203.219.99.194	v
9.				x
10.				x
11.				x
12.				x

Fig. 2 DialPlan setup page

Click each index number to edit fields.

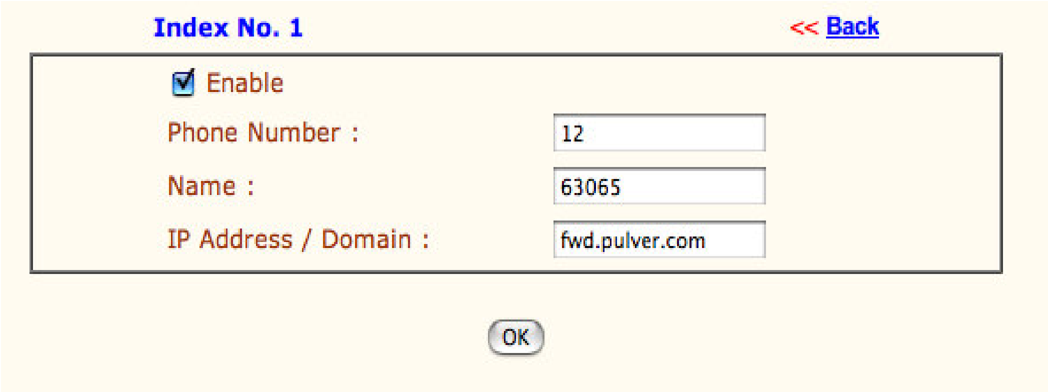


Fig. 3 DialPlan edit page

Enable: to enable this entry for mapping phone number when you dial the keypad on the phone.

Phone Number:a speed dial number.you can choose any number from 0~9 and*

Name: This field can fill with name(when using SIP protocol) or number.The number or name you filled must be the same as called party's setting.

IP Address / Domain: You can enter either IP address or domain name.

Example 1: if Tom give you a SIP URL as sip:63065@fwd.pulver.com then you can input the number just as Fig3, except you can change any number in the PhoneNumber field.

Example 2: if Aaron give you a sip url as sip:aaron@203.69.175.19 then you can enter the DialPlan as:

Phone Number: 1234 (any number you like)
Name: aaron
IP Address / Domain: 203.69.175.19

Example 3: if Calvin give you an IP address "203.69.175.16" only, and it is not in your dialplan, you still can use keypad on the phone to dial as **#203*69*175*19#**

Sip related function setup

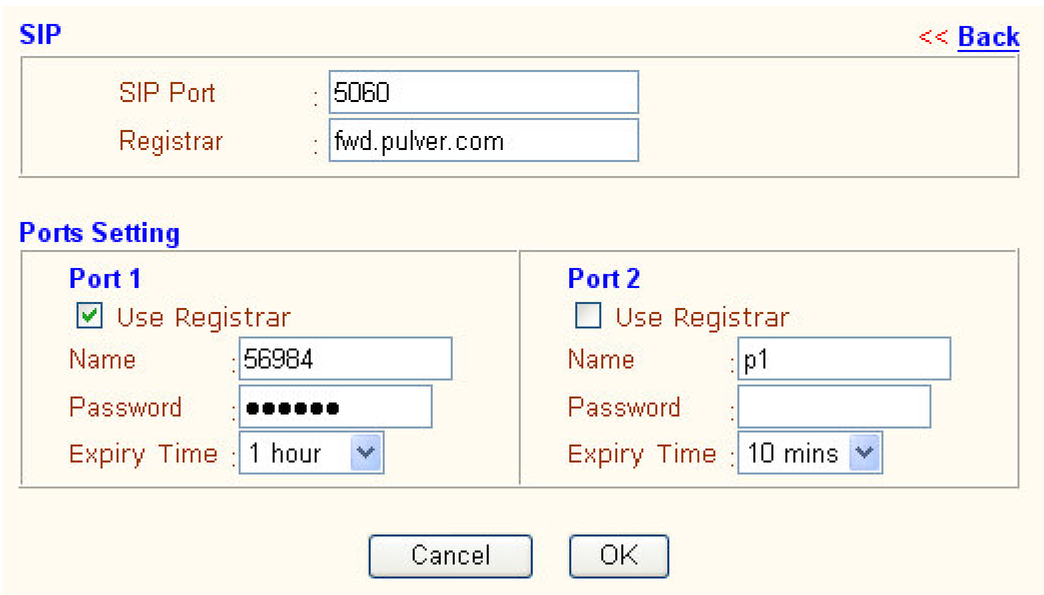


Fig. 4 SIP Related Function Setup Page

SIP Port: The port number is used to send/receive SIP message for building a session. While the default value is 5060, you can change it to other number. However, this situation needs other party to change simultaneously to the same number.

Registrar: you can enter domain name or IP address of SIP Registrar server. For example, iptel.org or 195.37.77.101 are identical. You have to apply an account of SIP Registrar server before you can use it.However, it is not necessary to use sip registrar server function in order to use VoIP function.

Use Register: check this box then you can use register function to register your Vigor with an SIP registrar server.

Name: You can enter a name or a number in this field. this field is the name part of SIP url.

Password: enter the password when you use a SIP registrar server which needs password.

Expire Time: The time duration that SIP registrar server keep your registration record. before the time expired, Vigor will issue another register message to registrar server again.

CODEC/RTP/DTMF Setup

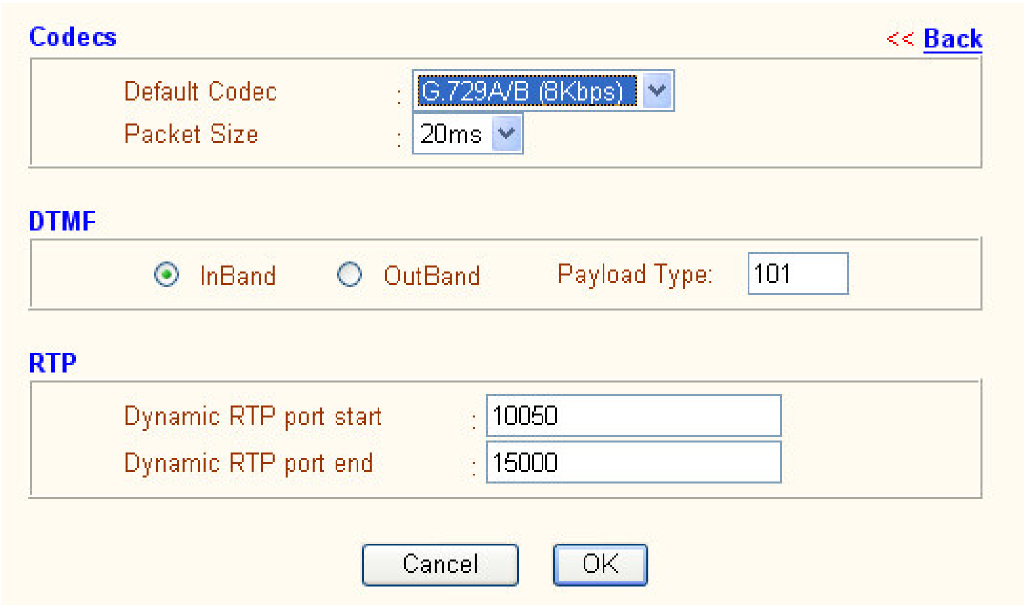


Fig. 5 CODEC/RTP/DTMF setup page.

Default Codec: there are five different CODECs you can choose as your prefer CODEC that you wish to use. However,the real CODEC be used was negotiate with peer party before session was established. The default CODEC is G.729A/B; it occupied less bandwidth while still have good voice quality.

NOTE: if your ADSL up stream speed only have 64Kbps, do not use G.711 CODEC.

Packet Size: the amount of data contains in a single packets. The default value is 20 ms, it means the data packet will contains 20 ms voice information. The more data contains in a single packet the less overhead it create but may increase .

DTMF InBand: Choose this one then the Vigor will send the DTMF tone as audio directly when you press the keypad on the phone

DTMF OutBand: Choose this one then the Vigor will capture the keypad number you pressed and transform it to digital form then send to the other side;the receiver will generate the tone according to the digital form it receive. This function is very useful when the network traffic congestion occur and it still can remain the accuracy of DTMF tone.

DTMF Payload type: Choose a number between 96 to 127, the default value was 101.

RTP: Specify the start and end port for RTP stream. The default values are 10050 and 15000.

Calling scenario

Peer to peer calling

There are two people ,say Kevin and Aaron. They both have 2600V in hand, so here's their settings in order to call each other.

Kevin's IP address is :214.61.172.53
Aaron's IP address is : 203.69.175.19

1. Kevin's setting:

1-1 DialPlan index 1

Phone Number: 1234 (any number you like)
Name: aaron
IP Address / Domain: 203.69.175.19

1-2 Sip related

SIP Port: 5060
Registrar: (leave blank, don't fill any thing)

Port 1:
Use Register:
Name:kevin
Password: (leave blank, don't fill any thing)
Expiry Time: use default value

1-3 CODEC/RTP/DTMF

Use default value.

2. Aaron's setting:

2-1 DialPlan index 1

Phone Number: 123 (any number you like)
Name: kevin
IP Address / Domain: 214.61.172.53

2-2 Sip related

SIP Port: 5060
Registrar: (leave blank, don't fill any thing)

Port 1:
Use Register:
Name:aaron
Password: (leave blank, don't fill any thing)
Expiry Time: use default value

2-3 CODEC/RTP/DTMF

Use default value.

3. Now, when Kevin wants to call Aaron, he picks up the phone and dials 1234.
4. When Aaron wants to call Kevin, he picks up the phone and dials 123

Calling via sip proxy Here's the scenario that two people call each other via sip proxy; this is a good way to calling when they use dynamic public IP addresses:Again, here's the setting for each other:

Kevin's sip url is : ***sip:kevinyu@iptel.org***
Irene's sip url is: ***sip:irene@iptel.org***

1. Kevin's setting:
   1-1 DialPlan index 1

      Phone Number: 611 (any number you like)
      Name: irene
      IP Address / Domain: iptel.org

   1-2 Sip related

      SIP Port: 5060
      Registrar: iptel.org
      Port 1:
      Use Register:V
      Name:kevinyu
      Password: ******** (enter the password)
      Expiry Time: use default value

   1-3 CODEC/RTP/DTMF

      Use default value.

2. Irnen's setting:

   2-1 DialPlan index 1
      Phone Number: 217 (any number you like)
      Name: kevinyu
      IP Address / Domain: iptel.org

   2-2 Sip related
      SIP Port: 5060
      Registrar:iptel.org

      Port 1:
      Use Register:V
      Name:irene
      Password:******** (enter the password)
      Expiry Time: use default value

   2-3 CODEC/RTP/DTMF
      Use default value.

3. Now, when Kevin wants to call Irene, he picks up the phone and dial 611.
4. When Irene wants to call Kevin, she picks up the phone and dials 217