

## Draytek Vigor 2910 Series

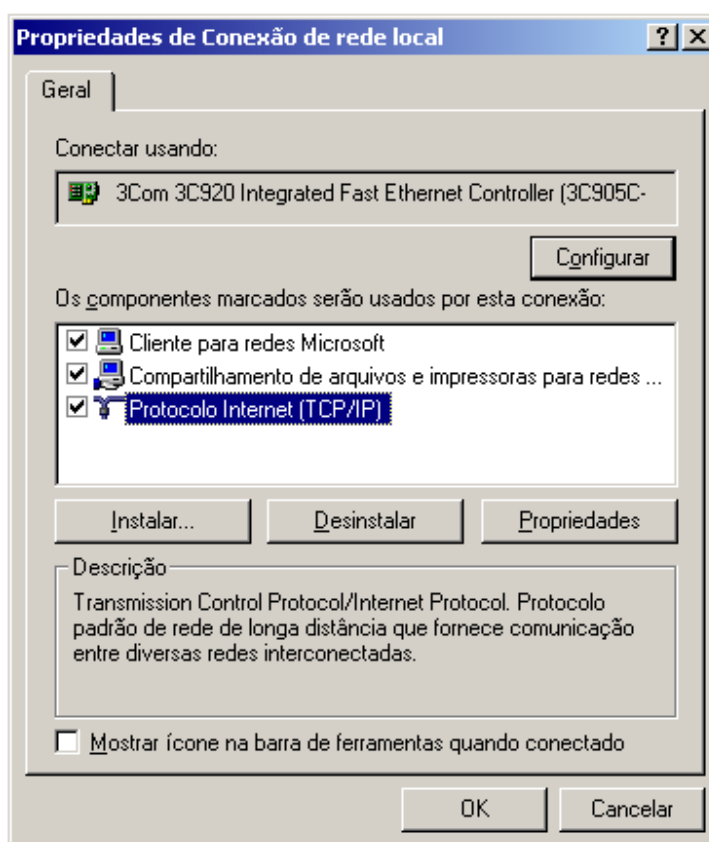
Configurando o computador para acessar as configurações do equipamento:

Você deverá deixar o seu computador conectado na porta P1 até P4 do seu **Vigor 2910 Series** e o mesmo deverá oferecer o IP automaticamente.

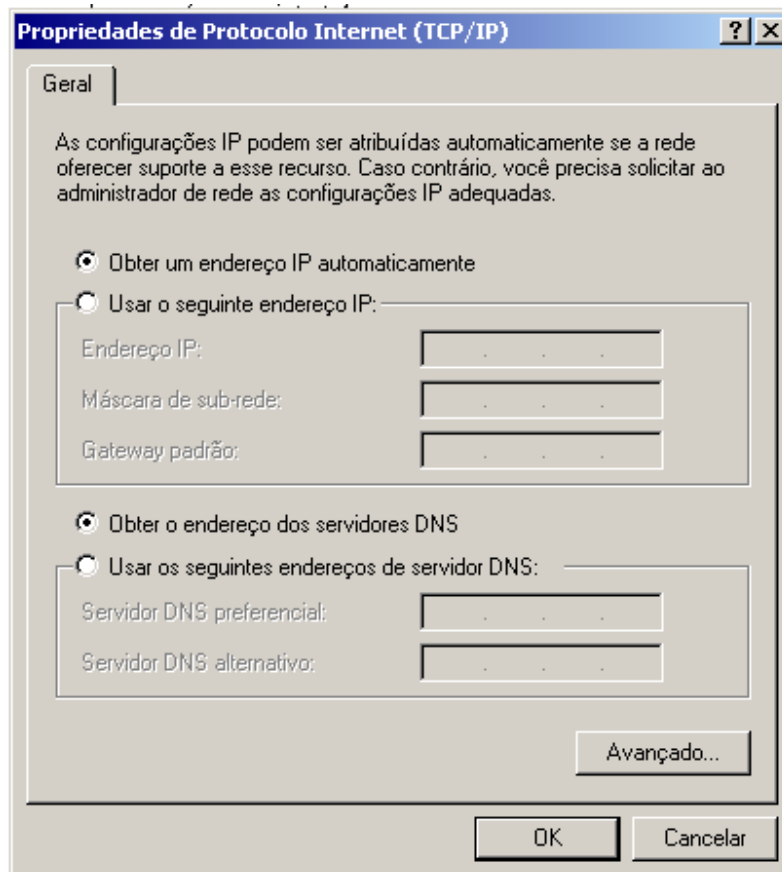
Para realizar a configuração para que o computador ofereça o IP automaticamente, siga os passos abaixo:

- Clique em Iniciar -> Configurações -> Painel de Controle;
- Abra a pasta Conexões dial-up e de rede;
- Clique com o segundo botão do mouse (geralmente o da direita) em Conexão de rede local e logo após clique com o primeiro botão (geralmente o da esquerda) em propriedades;

Aparecerá a seguinte tela:



Clique em cima de Protocolo Internet (TCP/IP) e logo após em Propriedades e sinalize as opções Obter um endereço IP automaticamente e Obter o endereço dos servidores DNS, conforme a figura abaixo:



Para acessar as configurações do equipamento, insira o endereço **192.168.1.1** no Internet Explorer e clique em **OK**, conforme a figura abaixo:

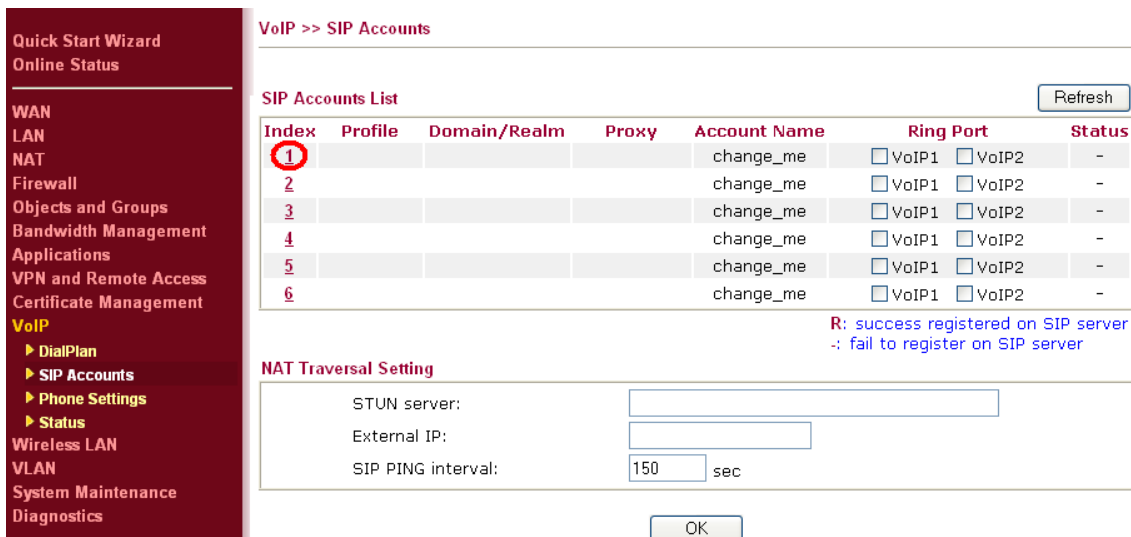


Ao acessar o endereço mencionado, aparecerá a seguinte mensagem:



- Basta clicar em **OK** (sem usuário e senha).

Clique em **VoIP** e logo após em **SIP Accounts**:



VoIP >> SIP Accounts

SIP Accounts List Refresh

Index	Profile	Domain/Realm	Proxy	Account Name	Ring Port	Status
1				change_me	<input type="checkbox"/> VoIP1 <input type="checkbox"/> VoIP2	-
2				change_me	<input type="checkbox"/> VoIP1 <input type="checkbox"/> VoIP2	-
3				change_me	<input type="checkbox"/> VoIP1 <input type="checkbox"/> VoIP2	-
4				change_me	<input type="checkbox"/> VoIP1 <input type="checkbox"/> VoIP2	-
5				change_me	<input type="checkbox"/> VoIP1 <input type="checkbox"/> VoIP2	-
6				change_me	<input type="checkbox"/> VoIP1 <input type="checkbox"/> VoIP2	-

R: success registered on SIP server  
 -: fail to register on SIP server

NAT Traversal Setting

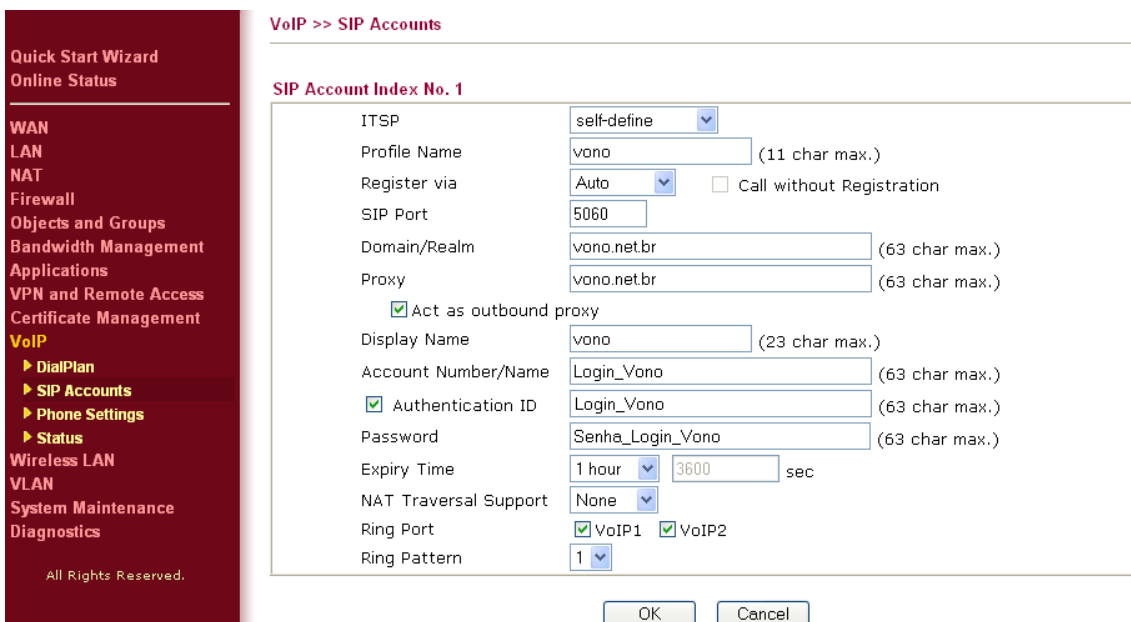
STUN server:

External IP:

SIP PING interval:  sec

OK

Clique no número **'1'** localizado abaixo do **Index**, conforme a figura mostrada.



VoIP >> SIP Accounts

SIP Account Index No. 1

ITSP: self-define

Profile Name: vono (11 char max.)

Register via: Auto  Call without Registration

SIP Port: 5060

Domain/Realm: vono.net.br (63 char max.)

Proxy: vono.net.br (63 char max.)

Act as outbound proxy

Display Name: vono (23 char max.)

Account Number/Name: Login\_Vono (63 char max.)

Authentication ID: Login\_Vono (63 char max.)

Password: Senha\_Login\_Vono (63 char max.)

Expiry Time: 1 hour  sec

NAT Traversal Support: None

Ring Port:  VoIP1  VoIP2

Ring Pattern: 1

OK Cancel

- Clique em **OK** e logo após em **OK** novamente;



Em “**Status**” você verá um “**R**” indicando que você está registrado com seu servidor SIP (Clicando em Refresh).

**VoIP >> SIP Accounts**

Quick Start Wizard  
Online Status

WAN  
LAN  
NAT  
Firewall  
Objects and Groups  
Bandwidth Management  
Applications  
VPN and Remote Access  
Certificate Management  
**VoIP**  
▶ DialPlan  
▶ SIP Accounts  
▶ Phone Settings  
▶ **Status**  
Wireless LAN  
VLAN  
System Maintenance  
Diagnostics

All Rights Reserved.

**SIP Accounts List** Refresh

Index	Profile	Domain/Realm	Proxy	Account Name	Ring Port		Status
<b>1</b>	vono	vono.net.br	vono.net.br	suportevono2	<input checked="" type="checkbox"/> VoIP1	<input checked="" type="checkbox"/> VoIP2	<b>R</b>
2				change_me	<input type="checkbox"/> VoIP1	<input type="checkbox"/> VoIP2	-
3				change_me	<input type="checkbox"/> VoIP1	<input type="checkbox"/> VoIP2	-
4				change_me	<input type="checkbox"/> VoIP1	<input type="checkbox"/> VoIP2	-
5				change_me	<input type="checkbox"/> VoIP1	<input type="checkbox"/> VoIP2	-
6				change_me	<input type="checkbox"/> VoIP1	<input type="checkbox"/> VoIP2	-

R: success registered on SIP server  
-: fail to register on SIP server

**NAT Traversal Setting**

STUN server:

External IP:

SIP PING interval:  sec

OK

Ainda no menu **VoIP**, clique em **Phone Settings**:

**VoIP >> Phone Settings**

Quick Start Wizard  
Online Status

WAN  
LAN  
NAT  
Firewall  
Objects and Groups  
Bandwidth Management  
Applications  
VPN and Remote Access  
Certificate Management  
**VoIP**  
▶ DialPlan  
▶ SIP Accounts  
▶ **Phone Settings**  
▶ Status  
Wireless LAN  
VLAN  
System Maintenance  
Diagnostics

**Phone List**

Index	Port	Call feature	Codec	Tone	Gain (Mic/Speaker)	Default SIP Account	DTMF Relay
<b>1</b>	FXS 1		G.729A/B	User Defined	5/5	vono	InBand
2	FXS 2		G.729A/B	User Defined	5/5	vono	InBand

**RTP**

Symmetric RTP

Dynamic RTP port start:

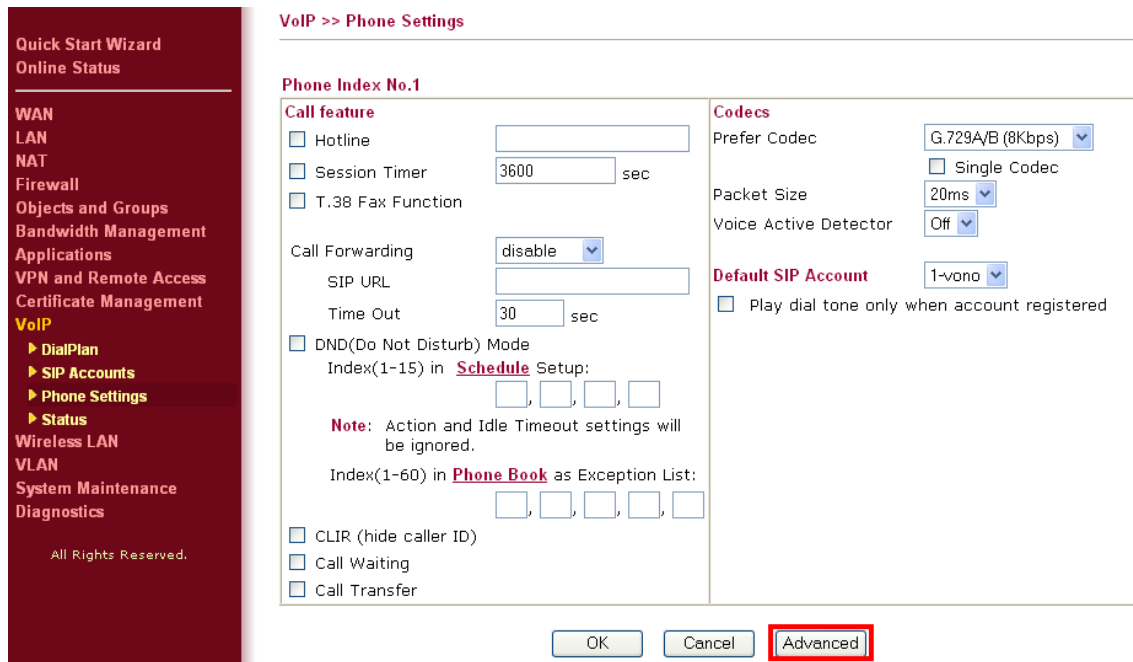
Dynamic RTP port end:

RTP TOS:

OK

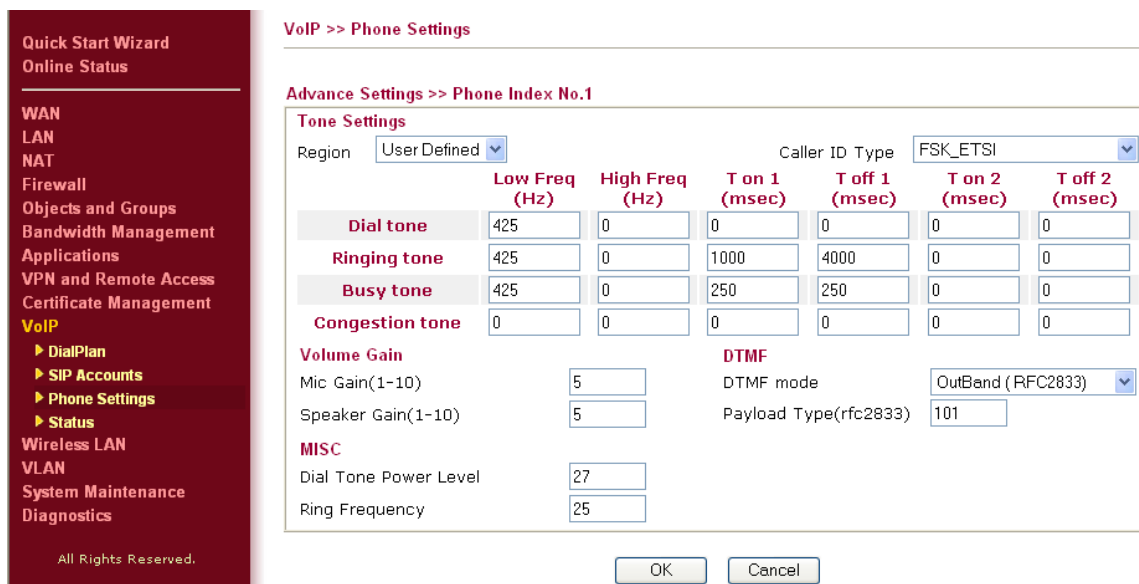
Clique no número ‘**1**’ localizado abaixo do **Index**, conforme a figura mostrada.

Copie a configuração proposta na figura abaixo e logo após clique em **Advanced** (marcado em vermelho):



The screenshot shows the 'VoIP >> Phone Settings' configuration window. On the left is a dark red sidebar menu with the following items: Quick Start Wizard, Online Status, WAN, LAN, NAT, Firewall, Objects and Groups, Bandwidth Management, Applications, VPN and Remote Access, Certificate Management, VoIP (expanded), DialPlan, SIP Accounts, Phone Settings (highlighted in red), Status, Wireless LAN, VLAN, System Maintenance, and Diagnostics. At the bottom of the sidebar is the text 'All Rights Reserved.' The main configuration area is titled 'Phone Index No.1' and is divided into two columns. The left column, 'Call feature', contains: Hotline (checkbox), Session Timer (3600 sec), T.38 Fax Function (checkbox), Call Forwarding (disable), SIP URL (text field), Time Out (30 sec), DND(Do Not Disturb) Mode (checkbox) with a note: 'Action and Idle Timeout settings will be ignored.', Index(1-60) in Phone Book as Exception List (checkboxes), CLIR (hide caller ID) (checkbox), Call Waiting (checkbox), and Call Transfer (checkbox). The right column, 'Codecs', contains: Prefer Codec (G.729A/B (8Kbps)), Single Codec (checkbox), Packet Size (20ms), Voice Active Detector (Off), and Default SIP Account (1-vono). Below the 'Default SIP Account' is a checkbox for 'Play dial tone only when account registered'. At the bottom of the window are three buttons: 'OK', 'Cancel', and 'Advanced' (highlighted with a red border).

Ao clicar em **Advanced**, copie as configurações:



**VoIP >> Phone Settings**

**Advance Settings >> Phone Index No.1**

**Tone Settings**

Region: User Defined (dropdown) Caller ID Type: FSK\_ETSI (dropdown)

	Low Freq (Hz)	High Freq (Hz)	T on 1 (msec)	T off 1 (msec)	T on 2 (msec)	T off 2 (msec)
Dial tone	425	0	0	0	0	0
Ringing tone	425	0	1000	4000	0	0
Busy tone	425	0	250	250	0	0
Congestion tone	0	0	0	0	0	0

**Volume Gain**

Mic Gain(1-10): 5 Speaker Gain(1-10): 5

**DTMF**

DTMF mode: OutBand (RFC2833) (dropdown)  
Payload Type(rfc2833): 101

**MISC**

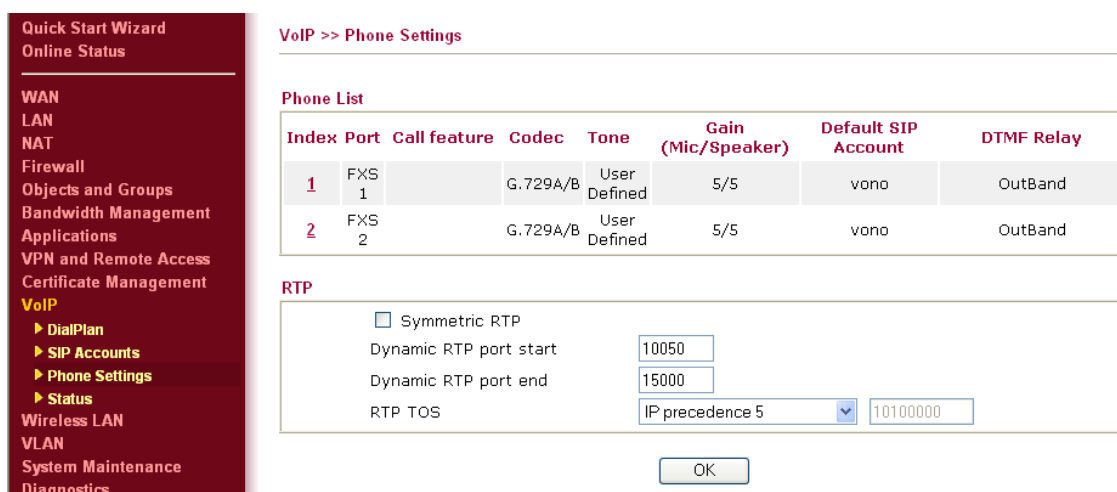
Dial Tone Power Level: 27  
Ring Frequency: 25

OK Cancel

- Clique em **OK** (3 vezes);

OBS: Faça esse item para o Index 2 desta vez;

Após aplicar as configurações no Index 1 e 2, as configurações deverão aparecer conforme a figura abaixo:



**VoIP >> Phone Settings**

**Phone List**

Index	Port	Call feature	Codec	Tone	Gain (Mic/Speaker)	Default SIP Account	DTMF Relay
1	FXS 1		G.729A/B	User Defined	5/5	vono	OutBand
2	FXS 2		G.729A/B	User Defined	5/5	vono	OutBand

**RTP**

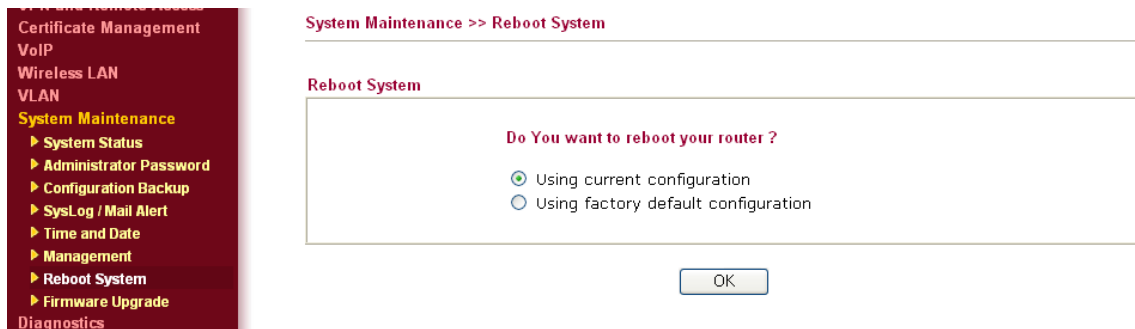
Symmetric RTP

Dynamic RTP port start: 10050  
Dynamic RTP port end: 15000  
RTP TOS: IP precedence 5 (dropdown) 10100000

OK



Clique em **System Maintenance** e logo após em **Reboot System**:



- Clique em **OK**;

Pronto, o seu equipamento está configurado para utilizar o serviço.