Siemens Gigaset C470IP: manual for a smooth installation

Providing professional solutions for your VoIP challenges

Difficulty	Speed	Comfort
2 / 10	*3 – 4 min.*	*Very good*

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Introduction

This manual was specifically designed in steps to give users an easy and quick understanding to establish their SIP account through the use of VoIPSolutions call minutes, also called SIP termination.

We don't offer you an alternative, but a full worthy solution for your phone bill!

This manual has a difficulty degree of **2** out of **10** on our VoIP-scale.

If you follow this manual step by step as pointed out, you should be able to configure this Siemens Gigaset C470 IP in about 3-4 minutes.

We base ourselves to our own provider service to fill in any concrete information according to this manual.

For all other information, except for configuring your SIP account and the most common problems you could encounter, you can find help here.

Before you go through this manual, I must point out that the pictures that are being used during this manual, have the title "Gigaset S450IP". This is a typo from Siemens Gigaset themselves, because this manual has without any doubt been configured using a Siemens Gigaset C470IP. Hopefully they will rectify this typo in a firmware update, this has been subtly referred to in STEP 2.

If you are already comfortable using this device and you only need specific information on how to configure your SIP account, then I would advise you to scroll down to **STEP 8**.

If you desire any further information or have any comments involving this product or this manual, please e-mail us on info@voipsolutions.be

General questions

Why a Siemens Gigaset C470IP?

The latest asset from SIP Siemens Gigaset products, are a really strong line of SIP devices that are the result of a happy marriage between SIP and DECT. This Gigaset also excels in an excellent sound quality with its Quality of Service label!

DECT is a wireless phone technology that has as strong points, a huge connection range and a long battery lifespan that will save you up to 60% on electricity. SIP on its turn brings out the best and also the cheapest in your phone bill!

Every Siemens Gigaset IP DECT device distinguishes itself through a great easy to use and user-friendly interface which of course results in having Siemens Gigaset to be in a prominent market position.

This Gigaset can naturally be connected with the Siemens Gigaset Repeater and also the Siemens Gigaset HC450, even when you can't get enough of 1handset, you can always purchase an additional handset (maximum 6 handsets per BASE) namely the Siemens Gigaset C47H, of course you can find all of these products on our web shop.

What offers the Siemens Gigaset C470IP more than the C450IP?

The most dramatic changes are that the Siemens Gigaset C470IP now has the possibility to save up to 6 SIP accounts and also has the possibility to store 150 names & numbers in the phonebook.

The stand-by time has been doubled from 150 to a staggering 300 hours and instead of 10 hours you can now benefit from up to 12 hours of conversation time.

This device of course offers a lot more extras than its predecessor, you can view all this information if you click here.

Why combine VoIPSolutions termination with a Siemens Gigaset C470IP?

As mentioned earlier, this manual has been specifically designed by VoIPSolutions for the use of a Siemens Gigaset C470 IP.

Associated with SIP termination services from VoIPSolutions, you can get a user-friendly and a cheap solution in your house that allows you to call freely and let's not forget very cheap!

By purchasing VoIPSolutions termination, you can configure your device to be ready for use in about 4 minutes by using this manual.

Naturally you can purchase this termination on our web shop. Here you can also see our rates to give you an easy overview of our prices.

This device is being used by ourselves because of its user-friendly interface and easy to configure web configuration.

How do I order SIP termination?

- As mentioned earlier you can order SIP termination on our web shop through this link or you can go to our web shop and click on the left side of your screen in the summary on "*VoIP Termination (Buy CallTime)*".

Before you decide to buy SIP termination you first have to be logged into the web shop or you can create a new account.

- Here you get a summary of 4 possible packets that we offer to our clients.

We offer you the possibility between 5 EUR, 25 EUR, 50 EUR and 250 EUR.

- You can immediately click on "*BUY NOW*" or you can first select the kind of termination you want to buy and then scroll down to the button which states "*ADD TO CART*".
- When you have completed this step, you find yourself looking at the products in your cart. If you wish to buy other products also then you simply click on the button "*CONTINUE SHOPPING*". If you have everything you need, you just press the button "*CHECKOUT*".
- Now you arrive at the payment information page. Here you can modify your shipping address or billing address. Furthermore you also have the possibility to choose a payment method, of course ordering SIP termination happens online so there are no shipping costs. There is also a possibility to send us some information about your order or place a comment about your order. When you are done with this you just have to press the button "*CONTINUE*" on the bottom of your page.
- Here you find the final step in your order process namely the "*ORDER CONFIRMATION*". Careful, your order has not been processed yet! Here you can check all the information that you entered and make changes to incorrect information by pressing the "*EDIT*" buttons that are spread in the order confirmation page. Now you just have to click "*CONFIRM*" on the bottom of the page to finally confirm your order.

We strive towards sending your login information within the hour of payment during office hours!

Configuring SIP account

Step 1 – How to connect to my base

How do I connect to my base?

- You take your handset and go to *MENU*
- You scroll down and when you see *SETTINGS*, you press *OK*
- Again scroll down until you see *BASE*, then press *OK*
- Scroll down until *LOCAL NETWORK* and press *OK*
- Dial in the *PIN* code (if you haven't already put in your own personal PIN code, then the standard code is "**0000**", without the quotation marks of course) and you press *OK*
- There you can see the *IP ADDRESS* that you need to connect to your base(e.a. 010.011.000.163)

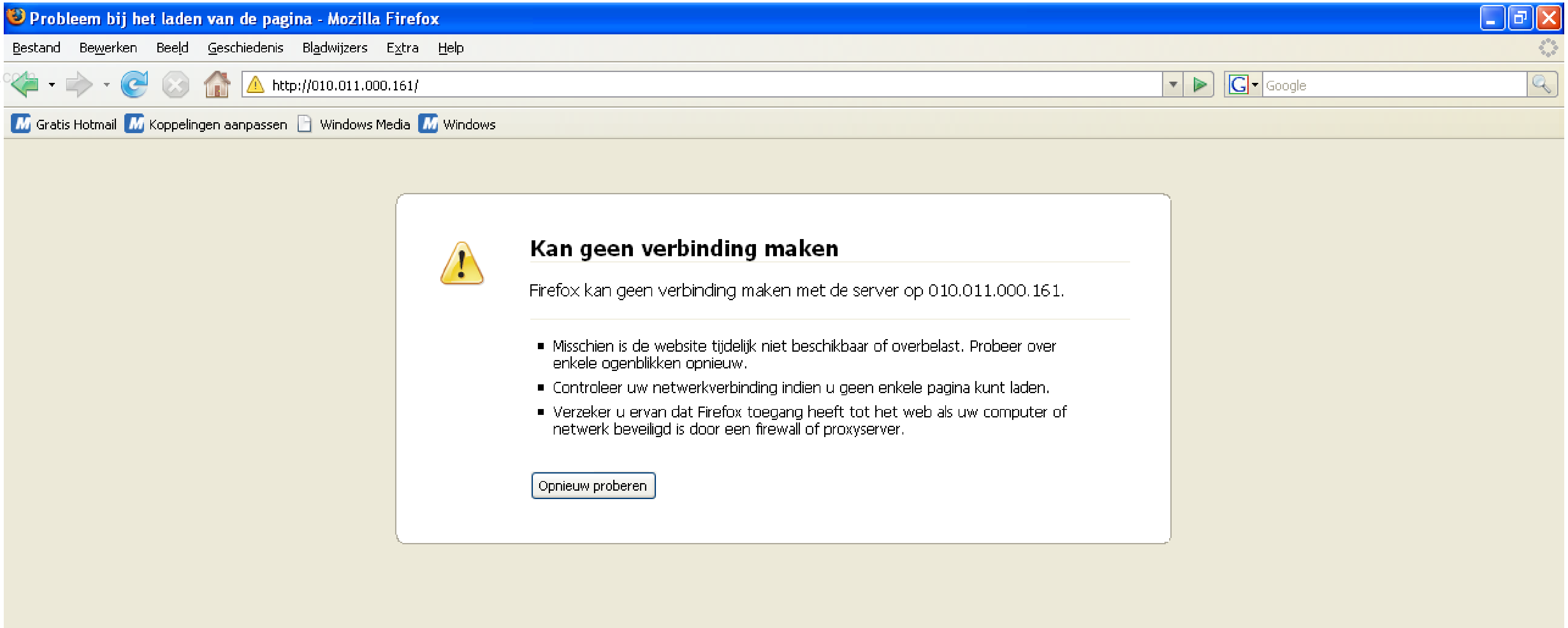
For more information about our products, stock or technical questions you can always contact us via e-mail : info@voipsolutions.be
Tel. : +32-2-7470777 (BE)
+31-20-2629991 (NL)

Step 2 – Putting in the IP Address

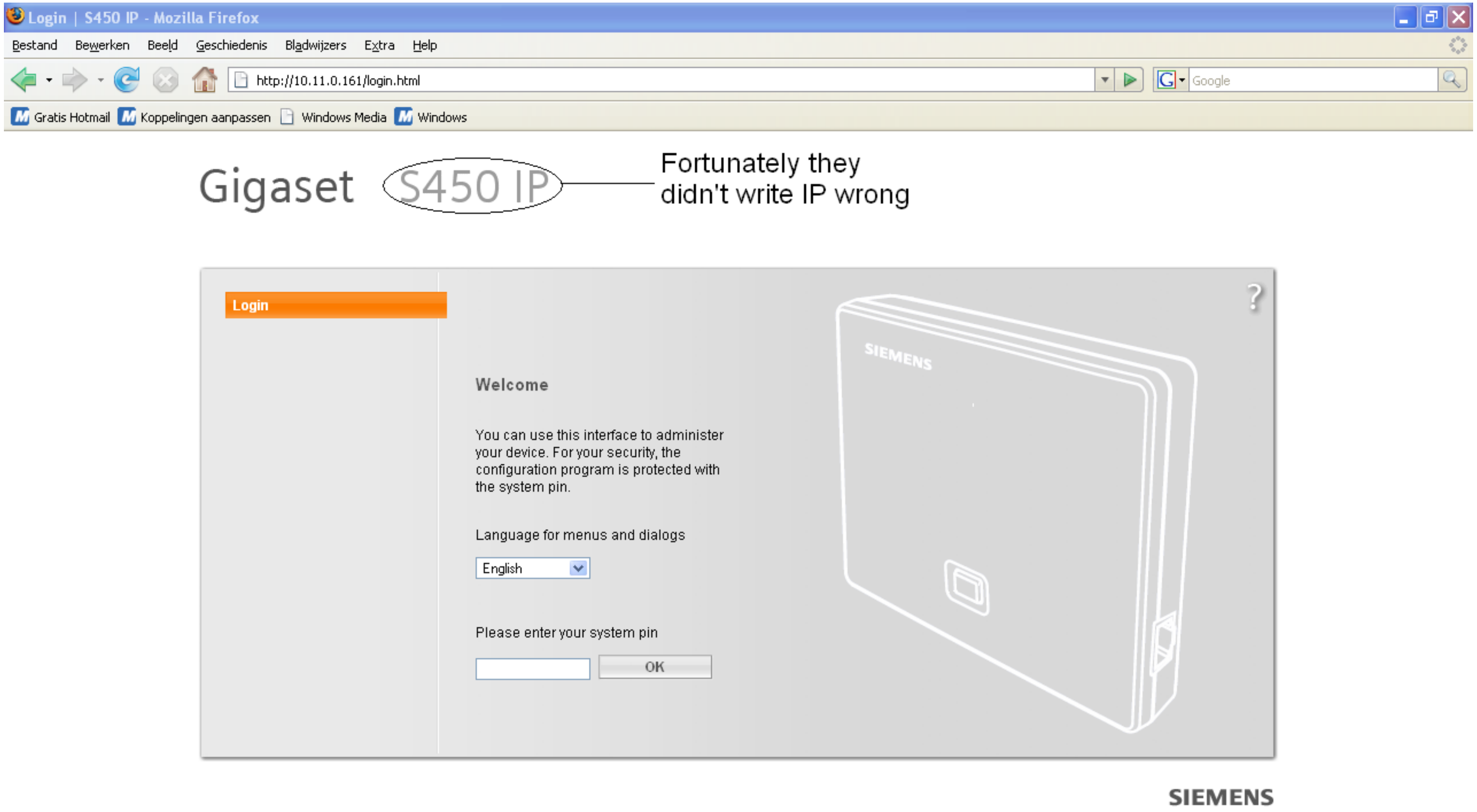
- You open your web browser and type in the IP ADDRESS in the place where you normally see the *http* address from a website.

!!! CAREFUL !!!

NO:

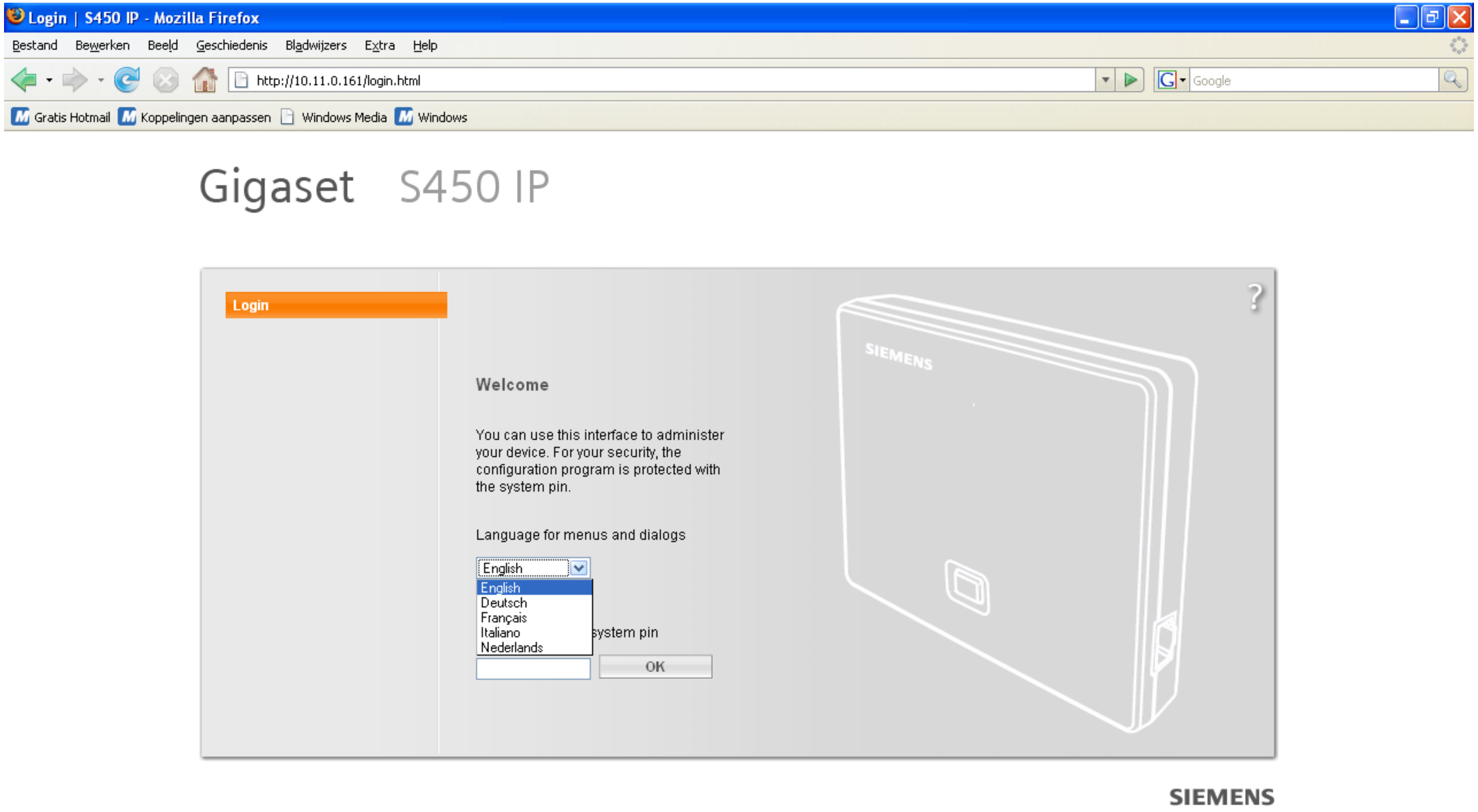


YES:



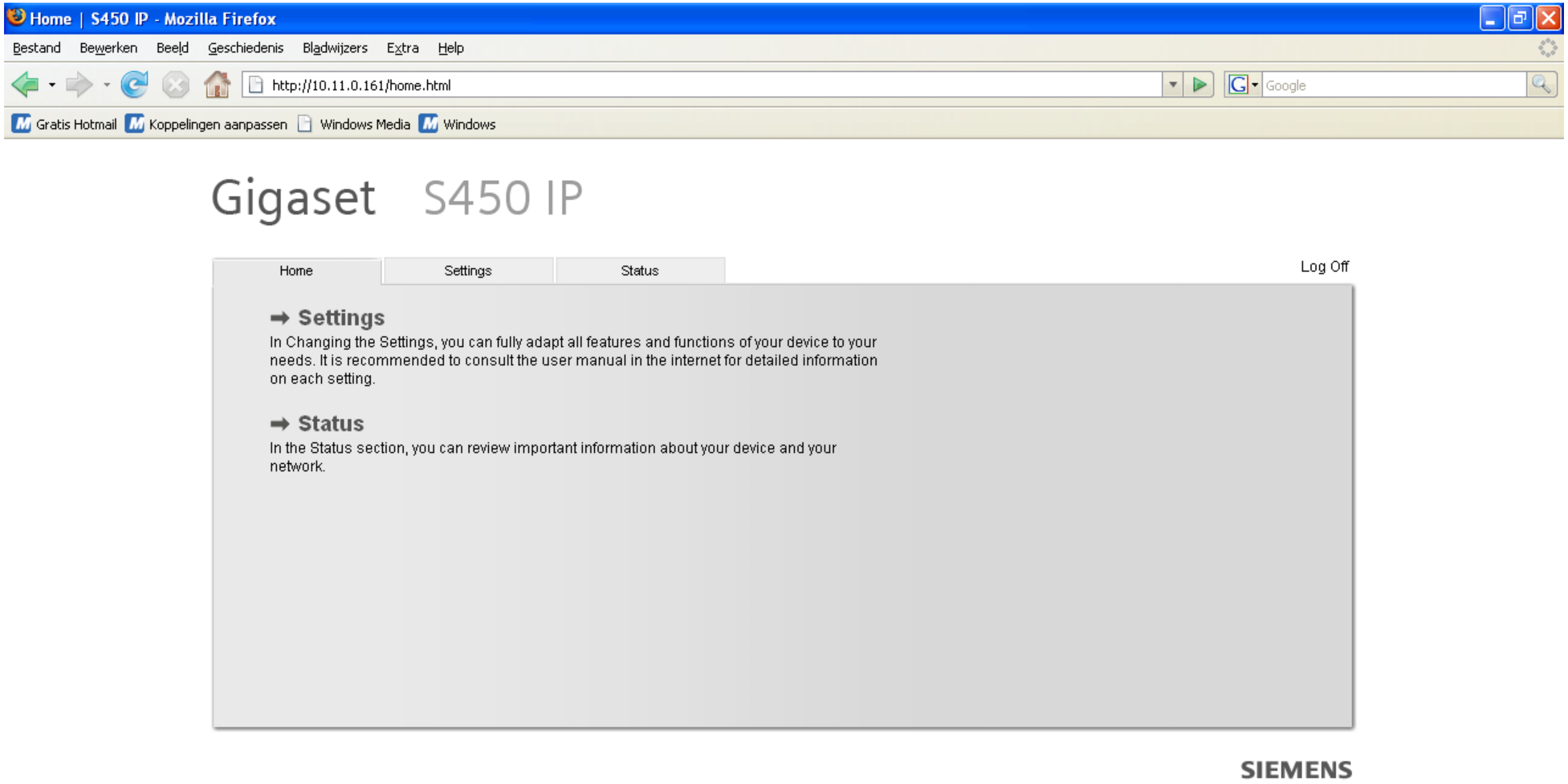
- The IP ADDRESS that you were give from the handset is in this example 010.011.000.161, but must be written in the webbrowser as 10.11.0.161.

Step 2.1 – Language preferences



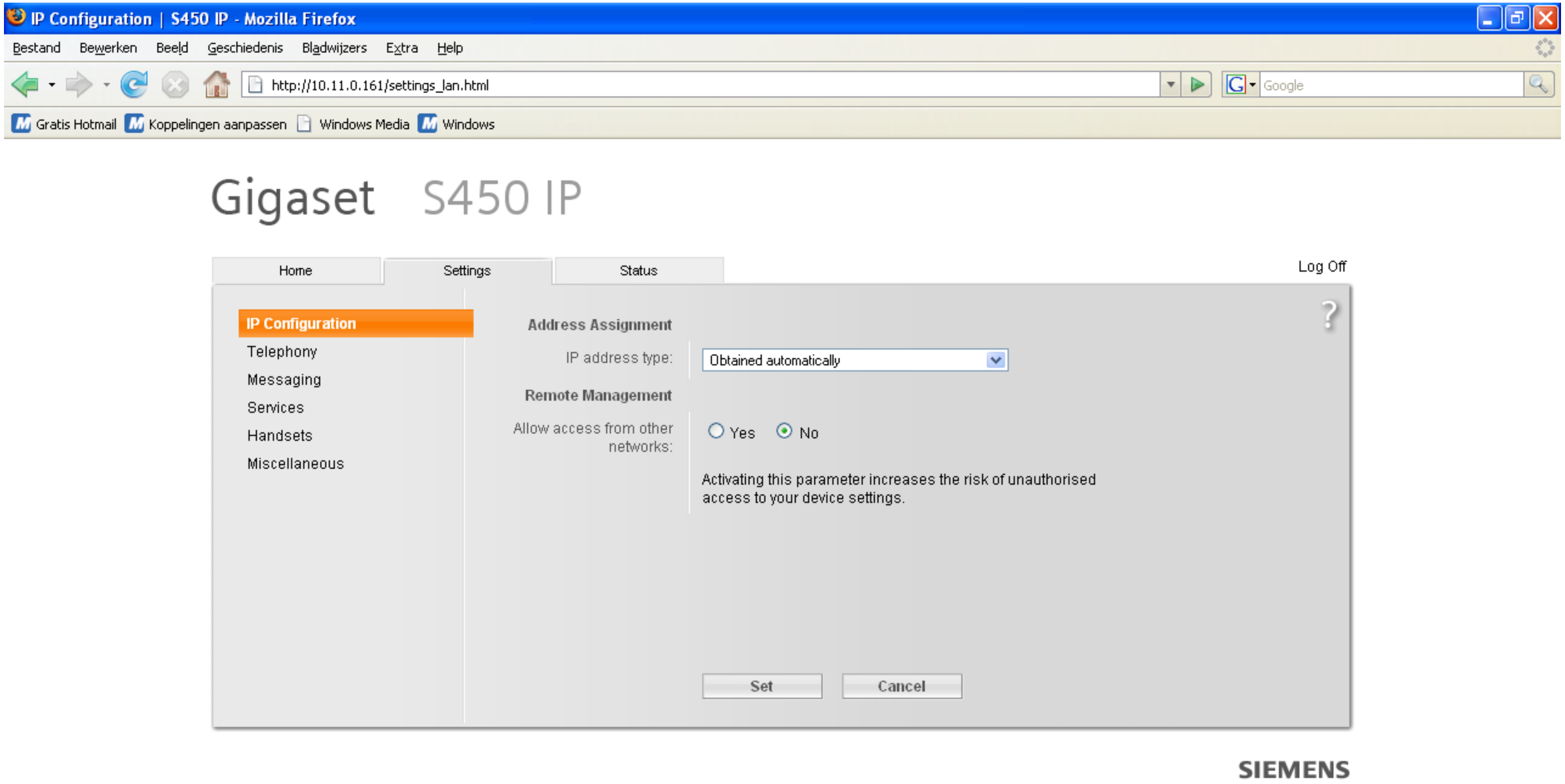
- Here you can determine your language preferences. This manual goes on according to the English language preference.

Step 3 – Configuration menu



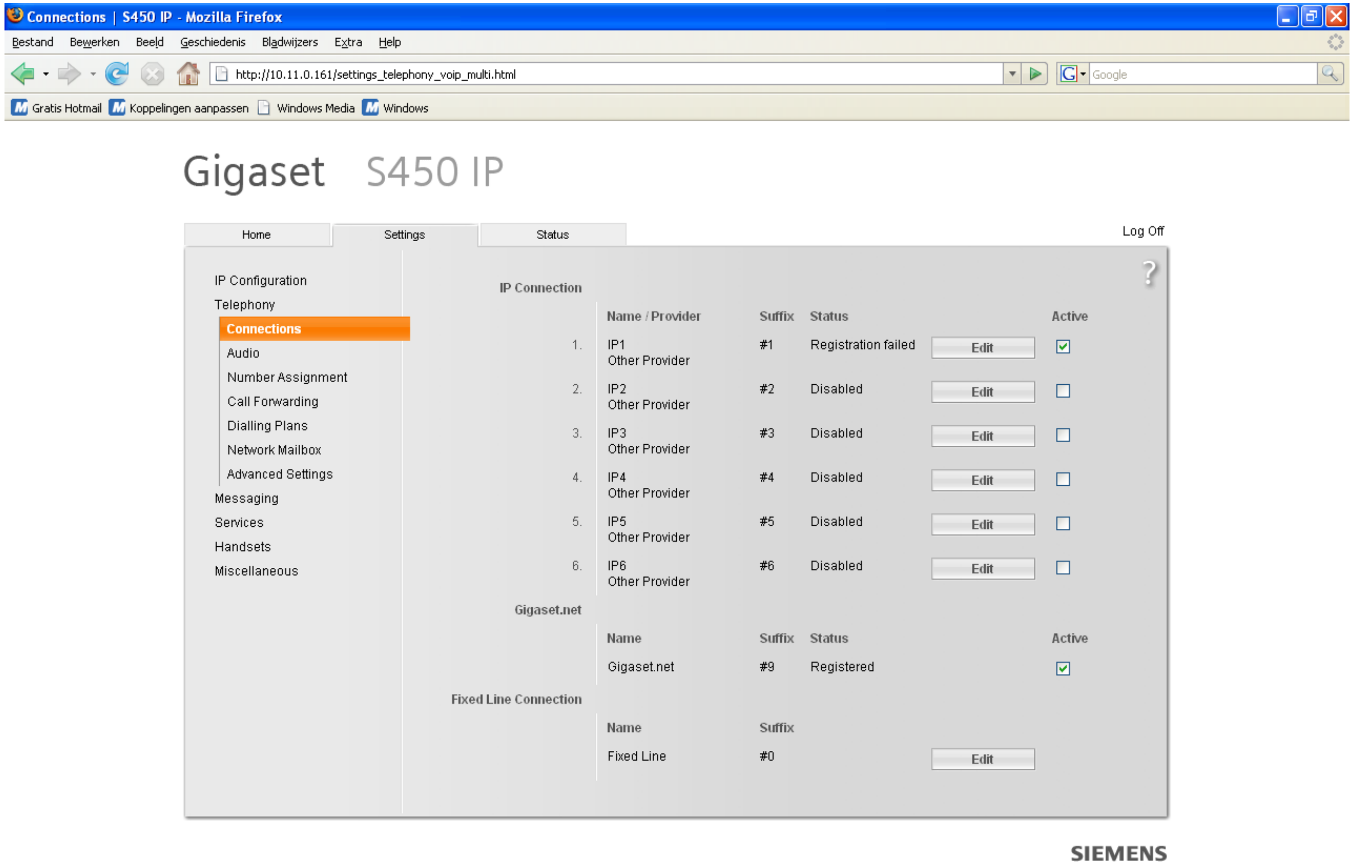
- Once you are logged in, you find yourself looking at the main menu of your BASE.
- Click on *SETTINGS*.

Step 4 – Settings local network



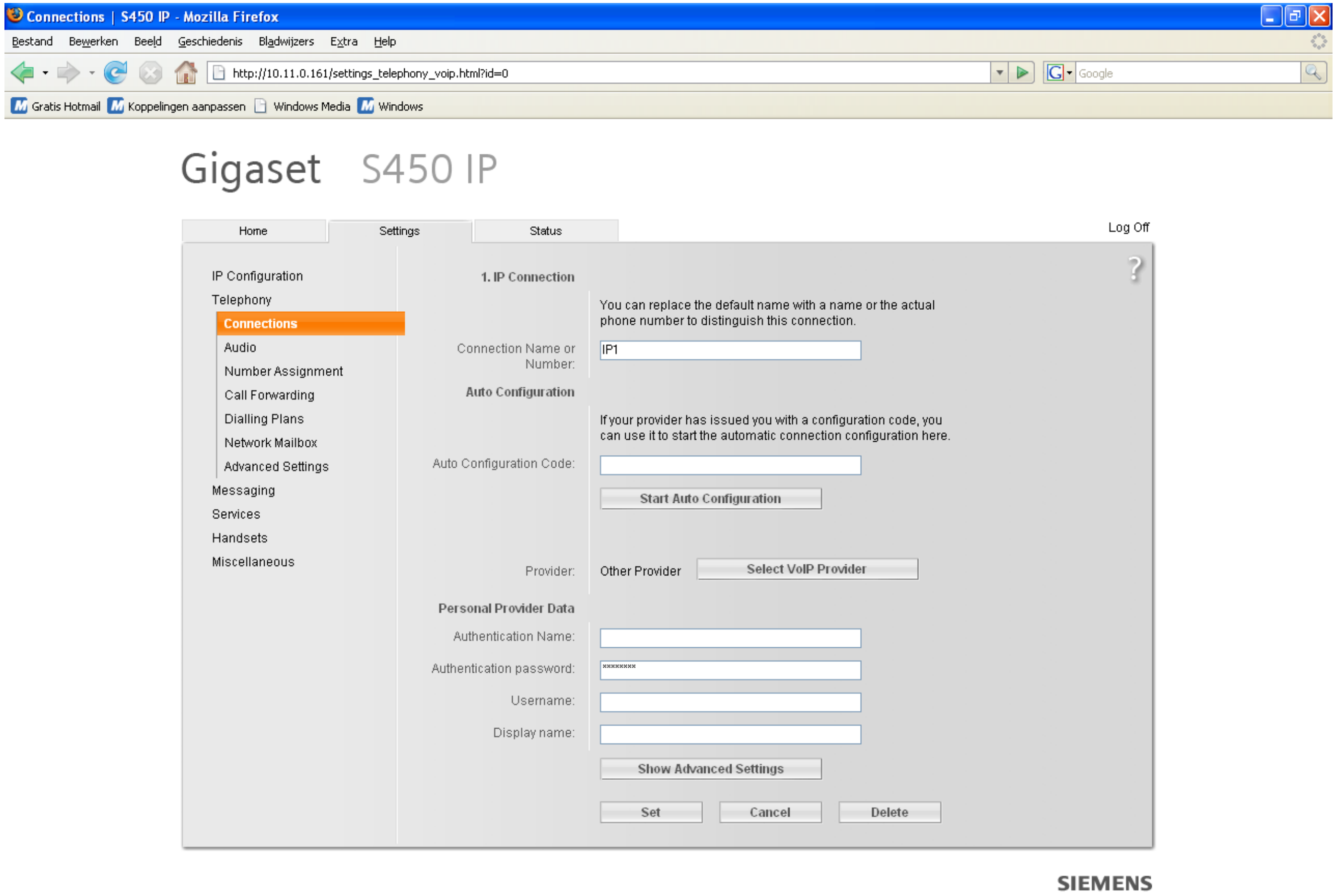
- Once you followed the previous steps, you should find yourself in the *SETTINGS* main menu.
- Now you click on "*TELEPHONY*" on the left side of the main menu.

Step 5 – Settings connections



- Here you get some subcategories under the category *TELEPHONY*, the one that is vital for you to configure your SIP account is the subcategory *CONNECTIONS*.
- Now you just have to click on the button *EDIT* that's located next to the 1e IP-connection, as the picture shows we are talkin about IP1 which states "Registation failed!".

Step 6 – Settings advanced options

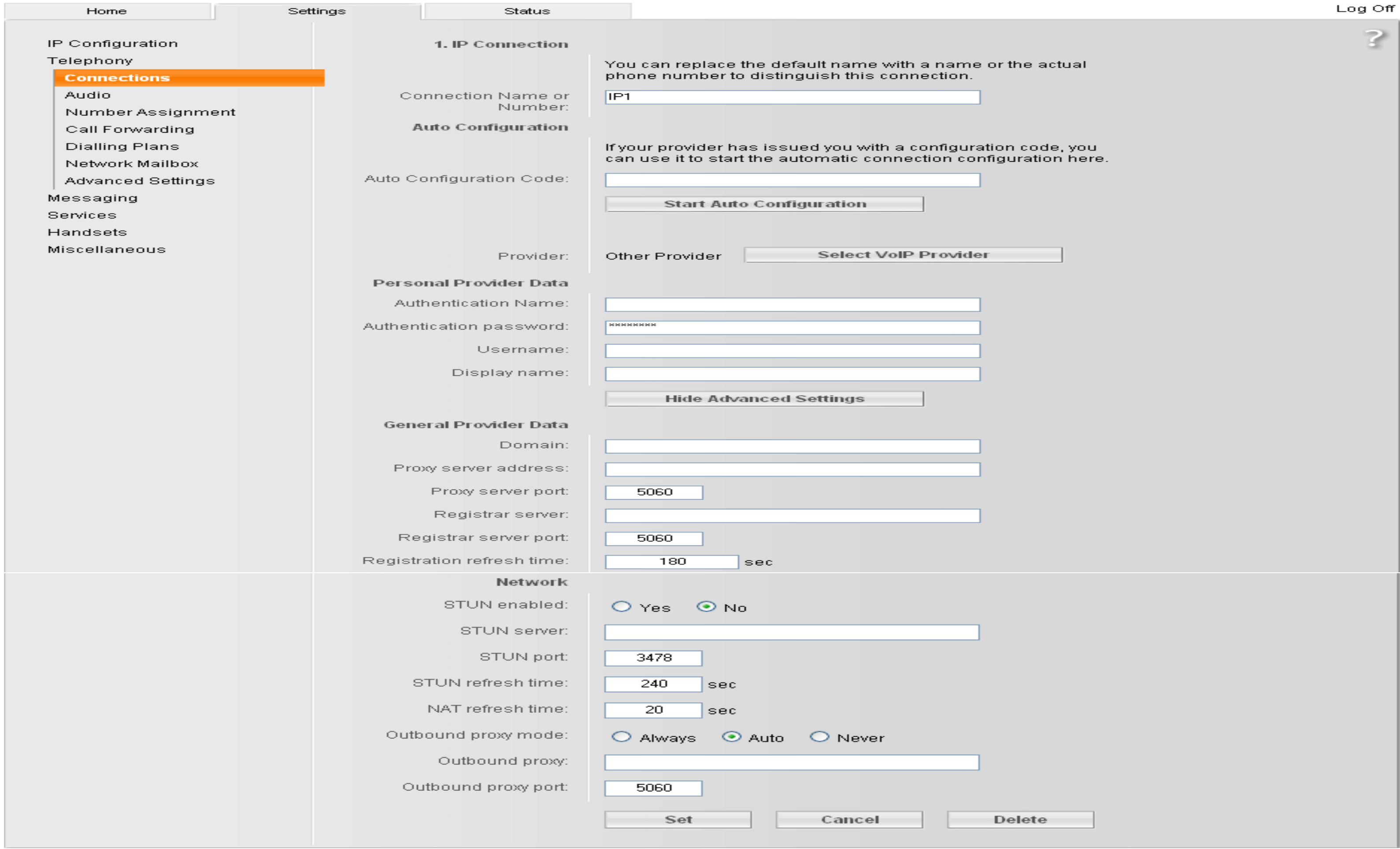


- You are now located in the menu to change the configuration of your SIP account.

- Please click on the button "*SHOW ADVANCED SETTINGS*" to fill in all necessary information that your SIP account needs in order to function normally. This button is located above the button "*SET*" at the bottom of the picture.

For more information about our products, stock or technical questions you can always contact us via e-mail : info@voipsolutions.be
Tel. : +32-2-7470777 (BE)
+31-20-2629991 (NL)

Step 7 – Settings advanced options blank



- Here you can fill in the information that you have been given from your provider for your SIP account

*Please start reading from **STEP 9** for a concrete filling-in of this information.*

Step 8 – SIP termination information

- When you have ordered Call Minutes at our web shop you will get, after confirmation, an e-mail stating a link to where you can view your SIP account information and also your call balance. You can view this information here.
- Keep this information on close hand, since this is our starting point which we use to fill in all the information on your BASE.
- At **STEP 11** we'll be coming back to this picture to give you some more and especially useful information, like how you can view from where, to whom you have called using this menu and also how long you have called and how much it costed you.

My VoIP Account Details

Balance: EUR 31.80 Excl. VAT
Username: [redacted] 1 **Password:** [redacted] 2 **Server:** sip.voipsolutions.be 3

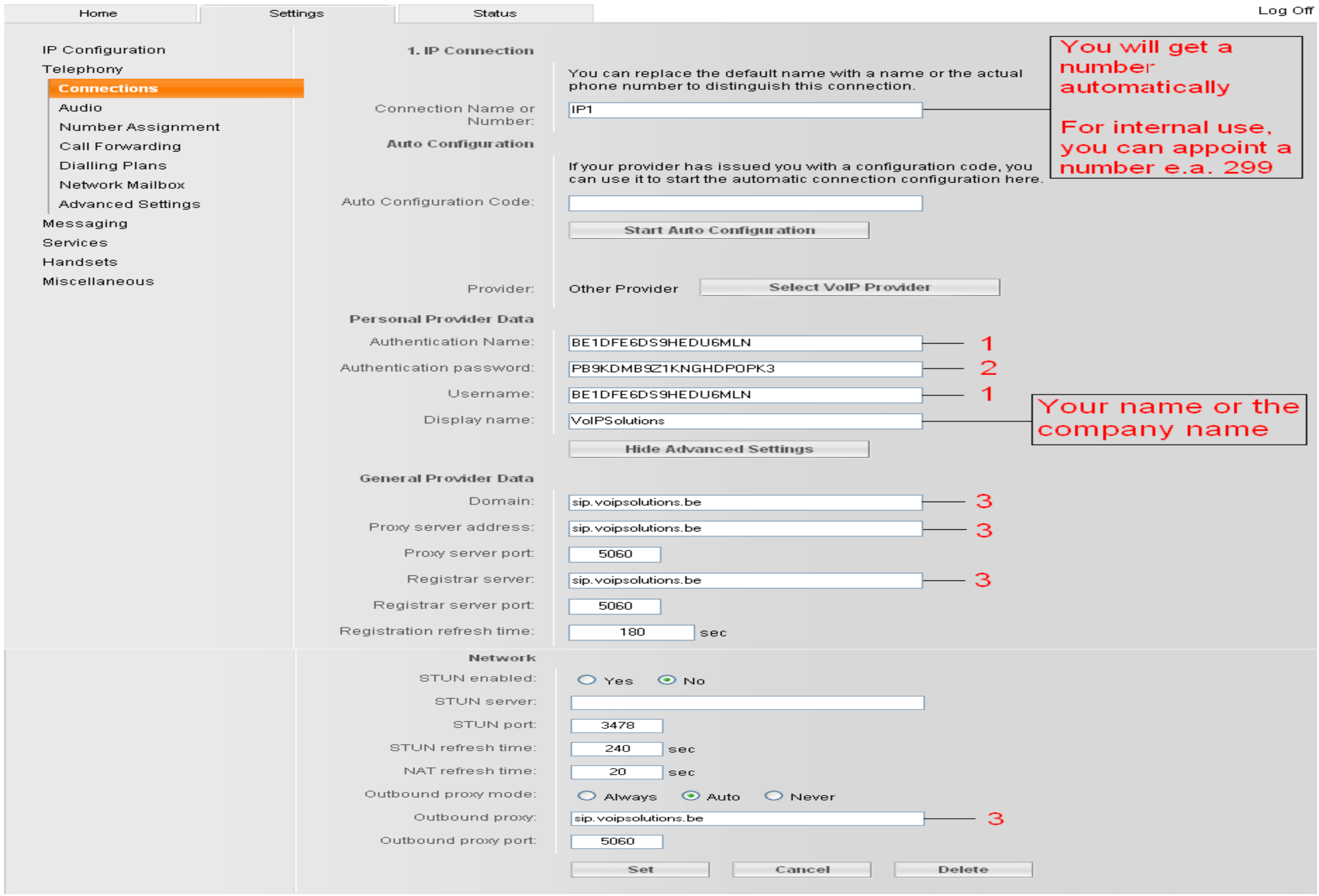
VoIP Orders

Order ID	Amount	Paid At	Status
90	41.32	2008-01-17 17:39	APPROVED

VoIP Call Reports

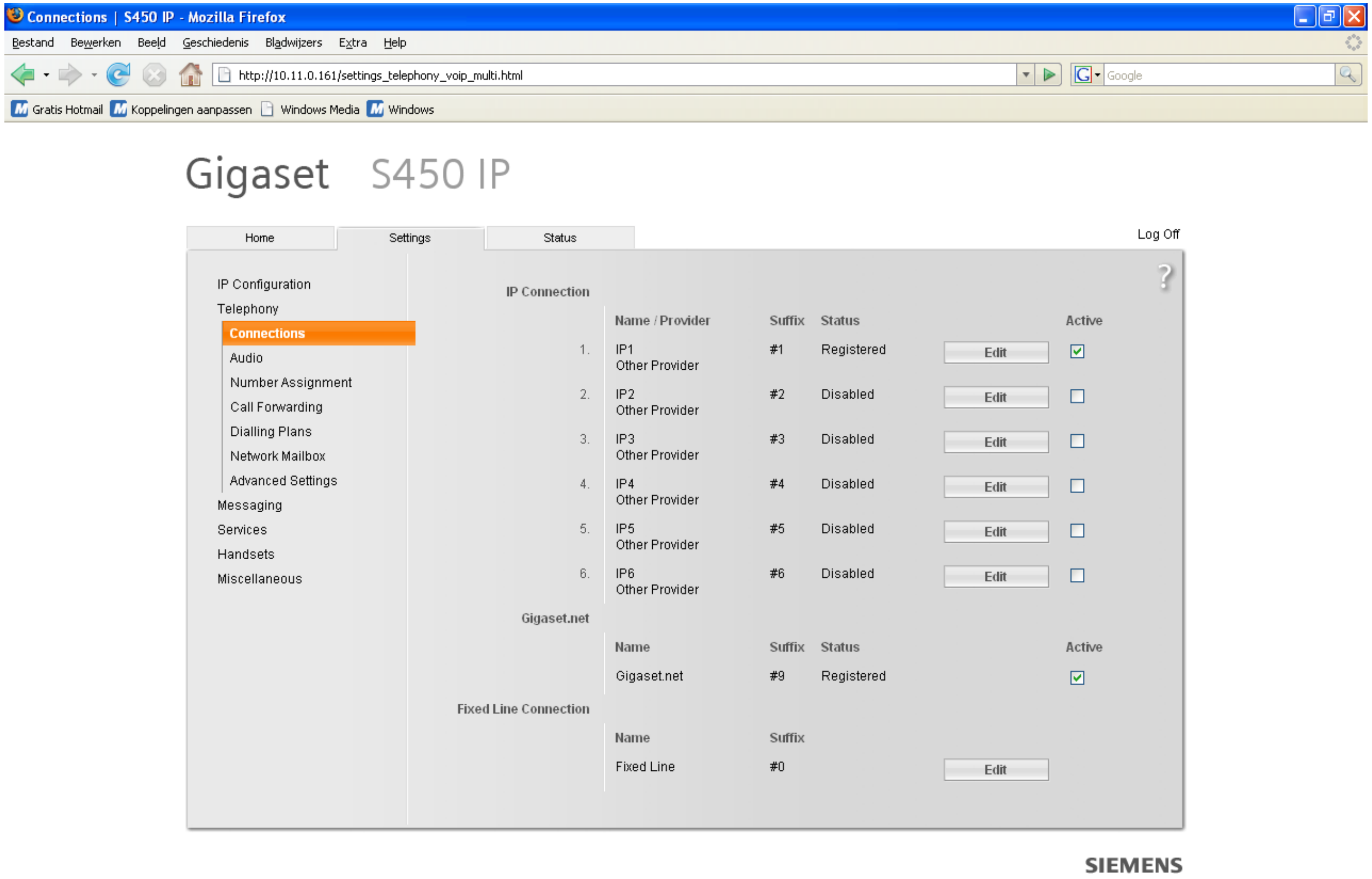
Download detailed call report for: November 2007 December 2007 January 2008

Step 9 – Filling in the settings according to SIP termination



- You have to fill in all information that is pointed out with a red text or a red number. The other settings are standard correctly filled in.
- After you have done this, just click on the bottom of the page on the button "*SET*".

Step 10 – Settings successful



- When you have correctly filled in the information of your SIP account and have saved this, you should return to this page as pointed out on **STEP 5** and you should see that the status has been changed to "*REGISTERED*".

- If the device still shows "Server is unavailable" or "Registration failed", please be patient for 5 to 10 seconds and try refreshing the page that is displayed on the picture.

Step 11 – Explanation SIP termination summary

My VoIP Account Details

Balance: EUR 31.80 Excl. VAT — 1
Username: [redacted] — 2 **Password:** [redacted] — 3 **Server:** sip.voipsolutions.be — 4

VoIP Orders

Order ID	Amount	Paid At	Status
90	41.32 — 5	2008-01-17 17:39	APPROVED

VoIP Call Reports

Download detailed call report for: November 2007 December 2007 January 2008 — 6

- Coming back to **STEP 8** we would like to offer you some further information about your SIP Account that is interesting for you.

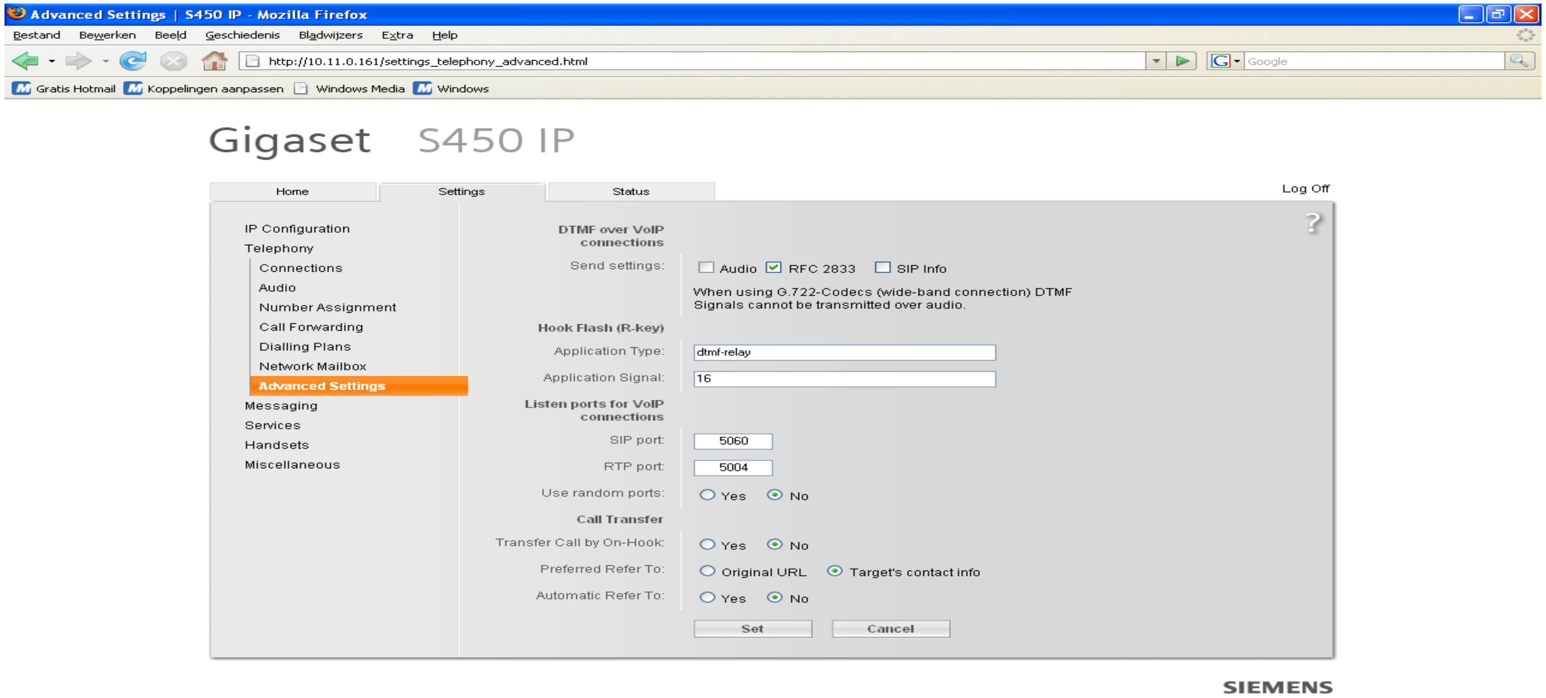
1. Remaining amount of call minutes (exclusive VAT)
2. Username (necessary at **STEP 9**).
3. Password (necessary at **STEP 9**).
4. Server (necessary at **STEP 9**).
5. Total amount of purchased call minutes (exclusive VAT).
6. Call history of all calls you have made, during the use of your SIP account.

Troubleshooting

DTMF

- DTMF problem, this stands for “Dual Tone Multi-Frequency” which is used for telephone signalling over the line in the voice-frequency band to the call switching center. The version of DTMF used for telephone tone dialing is known by the term Touch-Tone.

- Please execute the following step when the device isn’t responsive when you push a button or when your having difficulty forming the desired telephone number.



- Go to the category *TELEPHONY* and click on the subcategory *ADVANCED SETTINGS*.
- Check the box with the text “*RFC 2833*” **on** at the top of the page and check the box with the text “*AUDIO*” **off** and click on *SET*.

Registering SIP account failed

- Check if the network settings are correct (try to connect to your *BASE* as explained in **STEP 2**.
- Check the username and the password (space-sensitive en caps-sensitive)!
- Check if you have more than 1 SIP device on your network, if you have doubts try changing the local SIP port to another port than 5060.
    - For this you need to go back to the previous **DTMF**-page and change the port where the text says "*SIP-port*".

The handset won't register

- Reset your BASE and handset and let the handset find your base again, whilst holding the button on your BASE station pressed during some seconds and after these few seconds letting go of course (it's possible you will have to try this more than one time).

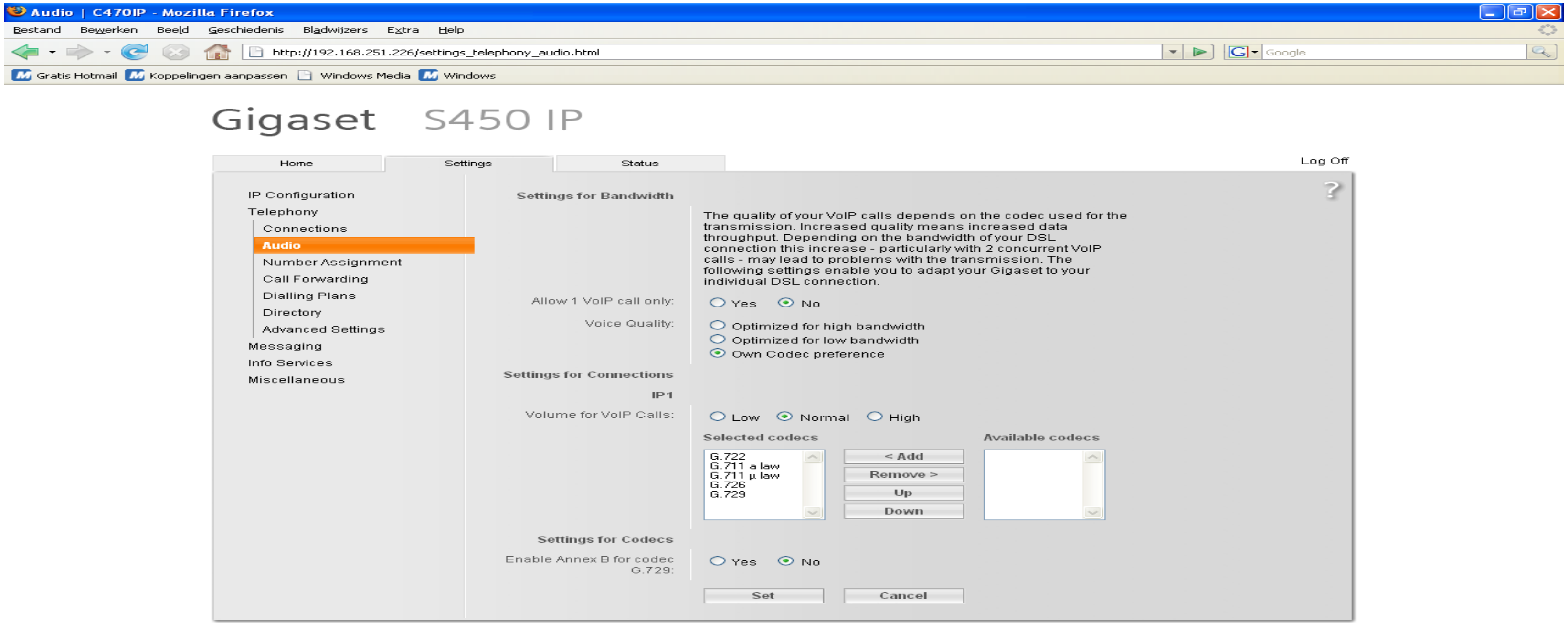
Outgoing calls successful, incoming calls sometimes

- Make sure that you don't have a double NAT (meaning a router behind a router).
- Decrease the NAT keepalive to less than 120 seconds.
- For this you need to go back to **STEP 9** and change the number in the box where the text says "*NAT refresh time*" to less than 120 seconds (this has been standard set at 20 sec.).

People complain that they don't understand me clearly

- You are using to little available bandwidth.
- Limit your outgoing traffic (think of Kazaa, Limewire, etc...).
- Use shaping on your router, to give VoIP priority on your network.
- Turn the volume of a VoIP conversation to the maximum.
- Verify if *CODEC G729* is being used, this has a warm sound and a low bandwidth.

- Please execute the following step if this codec isn't currently being used or when you are unsure about this.

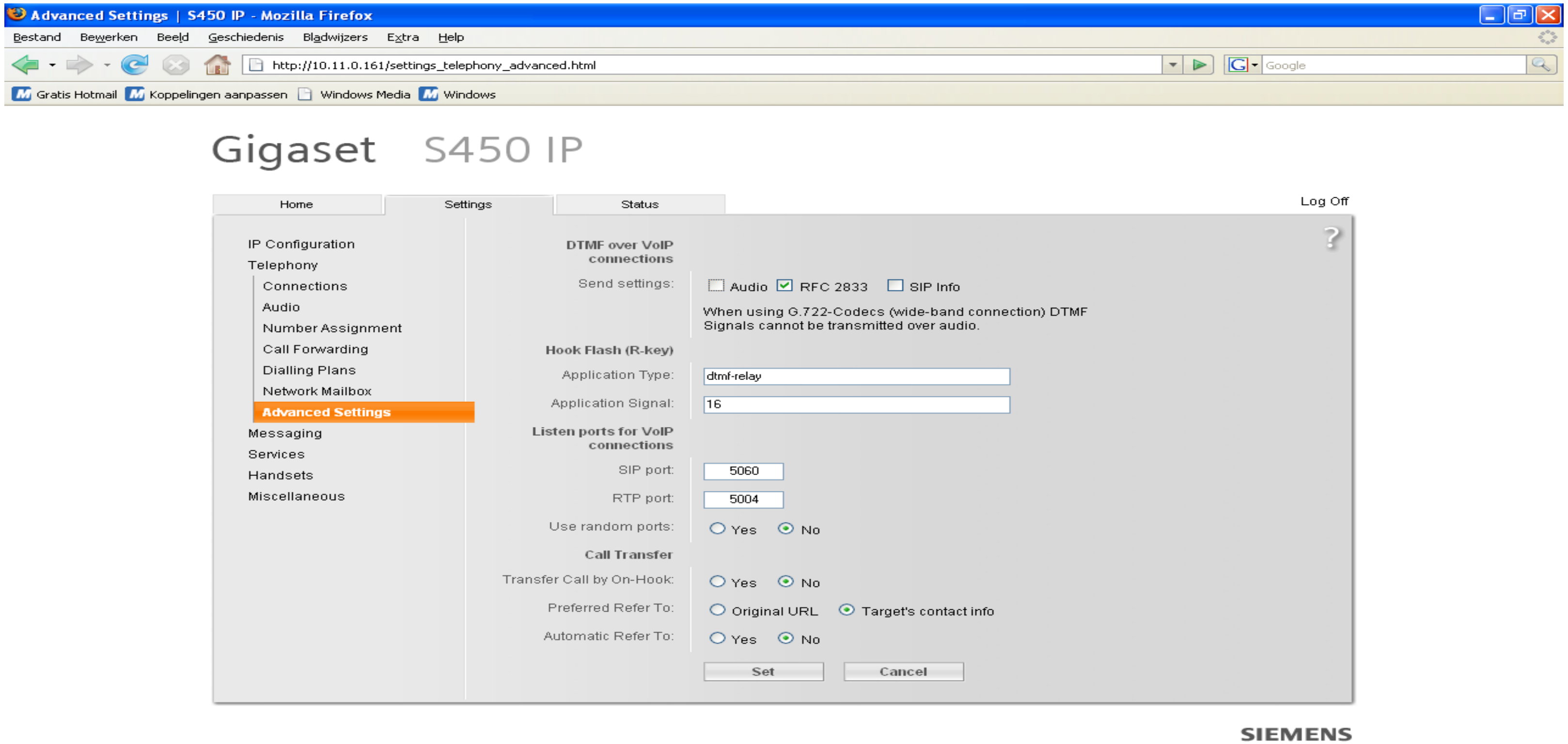


CAREFUL : You need to have your SIP account configured first before you see this screen!

- Go in the category *TELEPHONY* to the subcategory named *AUDIO*.
- As displayed should *CODEC G729* be in the list of *SELECTED CODECS* (this is set as standard).

People can hear me, but I can't hear them or vice versa

- You can also describe this problem as being "one-way sound".
- Avoid double NAT.
- Check if your firewall doesn't block the incoming traffic.
- Change the local SIP port.



Go to the same menu as displayed on the **DTMF**-page and change the *SIP-port*.