



Operating instructions

V101 / V102 SIP VoIP-Telephone Adaptor

English

Declaration of conformity and CE mark



This device meets the requirements of the following EC directive R&TTE 1999/5/EG:

»Directive 1999/5/EC of the European Parliament and of the Council of 9 March 1999 on radio equipment and telecommunications terminal equipment and the mutual recognition of their conformity«.

You can also request this EC declaration of conformity at the following Internet URL: <http://www.funkwerk-ec.com>.



The waste container symbol with the "X" through it on the device indicates that the device must be disposed of separately from normal domestic waste at an appropriate waste disposal facility at the end of its useful service life.

© 2007 Funkwerk Enterprise Communications GmbH - All rights reserved.

Reprinting of this document, even excerpts, is permitted only with the express consent of the publisher and with precise source information, regardless of the media used (mechanical or electronic).

Function descriptions included in this documentation which refer to software products of other manufacturers are based on the software used and valid at the date the documentation was prepared or published. The product and company names used in this documentation may be protected by trademarks.

Table of contents

Safety notes	1
Technical specifications	1
V101 SIP VoIP Telephony Adapter	2
V101 SIP VoIP Telephony Adapter Features	2
Compatibility	2
Package Contents	2
LED Indicators and Connectors	3
Installation / SIP Configuration	3
V102 SIP VoIP Telephony Adapter	4
V102 SIP VoIP Telephony Adapter Features	4
Compatibility t	4
Package Contents	4
LED Indicators and Connectors	5
Installation / SIP Configuration	5
Using Attached Analog Terminal Equipment	6
Symbols	6
Telephony	6
Configuration through the Web Browser (V101, V102)	9
System Information.	9
Phone Book.	10
Call Settings	11
Dial Plan Setting	15
Network.	18
SIP Settings.	21
Automatic Configuration Setup (Auto Config)	23
User Password	24
Save Changes.	25
Update	25
Default Settings	27
Reboot.	27
Configuration using the Telephone's Keypad	28
Application Examples	30
Troubleshooting the Web-based Configuration.	36
Short Reference Manual	37
Broadband Link through the TA (ADSL/cable modem)	37
ADSL links with external NAT router serving TA	37
Index	39

Safety notes

- Mind the storage and operation temperatures for the device listed in the technical specifications. Only connect the device when the permissible ambient operating temperature has been reached.
- Please note that condensation may occur on or in the unit when moving it from cold to warm surroundings. Only remove the unit from the package when the permissible ambient operating temperature has been reached.
- You should not make calls or connect or disconnect any lines during thunderstorms.
- Be sure to connect all cords and cables only to their designated ports.
- Be sure to route cords and cables so nobody can trip over them.
- Avoid the following:
 - Direct sunlight
 - Sources of heat (for example radiators)
 - Electronic devices (for example stereo components, office equipment or microwave devices)
 - Ingress of moisture or liquids
 - Aggressive fluids or vapors
 - Dusty environment.
- Do not use the unit in moist areas or where there is a risk of explosion.
- Do not expose any internal parts of the unit unless you are instructed to do so by the installation and operating manual.
- Do not touch any connector pins with sharp metallic items or damp objects.
- If the unit is not fixed to a surface, place it and its accessories on an anti-skid mat or equivalent.
- Clean the unit using only a slightly dampened piece of cloth.
- Do not use any accessories unless they have been explicitly approved.
- The unit may only be interfaced with terminal equipment that supplies SELV voltage (Safety Extra Low Voltage) and/or conforms with ETS 300047. This regulation is fulfilled when approved terminal devices are used as intended.

Power Supply Block

- Use only the power supply block supplied with the product or one that has been manufactured-approved as an accessory.
- Do not use any power supply block units with visible defects or damage (fissures, cracks in the enclosure)

Maintenance and Care

The V101 and V102 SIP VoIP telephony adapter is manufactured for normal conditions of use. When required, clean the ISDN system telephone with a slightly moistened cloth, or with an anti-static cloth. Never use a solvent to clean the device! Never use a dry cloth; electrostatic discharge may damage the electronics in the system. Under all circumstances make sure that no moisture penetrates into the enclosure and damages the SIP VoIP telephony adapter.

Installing the Telephony Adapter

Please note that the plastic feet of your SIP VoIP telephony adapter may leave dents in sensitive surfaces, for example on furniture. The manufacturer of the telephony adapter shall not accept any liability for such kind of damage. For that reason we recommend that you place your telephony adapter on an anti-skid surface.

Technical specifications

Power supply:	12V DC / 1A through 230 V AC power supply block supplied
Temperature range:	5° C to 40° C, 85 % max. humidity
Dimensions:	120 x 100 x 35 mm
Weight:	0.130 kg

V101 SIP VoIP Telephony Adapter

The V101 is a telephony adapter (TA) for »Voice over IP« applications (VoIP) based on SIP protocols. Once connected to the Internet the TA can link two stations having IP addresses with each other through the Internet. The TA features two Ethernet LAN ports for hooking up a DSL modem / router and a PC, as well as one port for an analog phone. Thanks to its integrated NAT/DHCP server the TA can be configured through a Web browser. This feature makes it ideally suited for customers of ITSPs (Internet Telephony Service Providers) and SOHO users, enabling them to use Internet telephony.

Each TA needs one IP address, one subnet mask and one Gateway / Router IP address for connecting to the Internet. This data is supplied to you by your Internet Service Provider. In case your provider uses dynamic IP address allocation towards their customers, users of the TA may activate the PPPoE or DHCP functions to have a dynamic IP address assigned automatically. The MAC hardware address (Media Access Control of TA) can be found on the bottom side of the TA. Additional information can be found on page 9 and beyond, »Configuration through the Web Browser«.

V101 SIP VoIP Telephony Adapter Features

The V101 TA offers the following features:

- SIP v1 (RFC2543), v2 (RFC3261) with MD5 Authentication (RFC2069 and RFC 2617).
- two RJ45 Ethernet ports and an RJ11 port for connection to analog terminal devices.
- ITU-T G.711, G.723, G.726, G.729A/B, VAD and CNG for Voice Codec.
- ITU-T G.165/168 Echo Compensation.
- Three LED Indicators: POWER, PHONE, LAN.
- Configuration supported through Web browser and keypad on phone set.
- Integrated NAT/DHCP server.
- PPPoE/DHCP Client for dynamic IP address allocation, plus NAT, DNS, and DDNS clients.
- Supports STUN Server for NAT Traversal.
- Hot Line Mode support.
- Supports T.38 FAX via IP.
- Interactive voice response system (IVR) reads out IP status of phone.
- Remote firmware upgrade from HTTP or TFTP server through Web-enabled PC.
- Direct IP/URL dialing without assistance through SIP proxy or subscriber number provisioning through SIP server.
- Telephony Features: Volume Adjustment, Phone Book, Speed Dialing, Call Forwarding, Call Waiting, Call Hold and Three-Party Conferencing, Redialing, and Flash.
- Outband DTMF (RFC 2833) / Inband DTMF / Send DTMF SIP Info.

Compatibility

The V101 VoIP TA supports the following standards:

VoIP Protocol:	IETF RFC3261 and RFC 2543 for SIP.
SIP Authentication:	IETF RFC2069 and RFC 2617 for MD5.
Voice codec:	see page 22, as well as VAD (Voice Activity Detection) and CNG.
Echo Compensation:	ITU-T G.165/168.

Package Contents

- V101 SIP VoIP Telephony Adapter.
- Power Supply Block 230V AC /12 VDC, 1 A.
- CD-ROM with User Manual.
- Brief description of functions.
- RJ 11 to TAE adapter.

Please check whether the packaging is damaged or whether parts are missing. If that is the case, please contact your dealer.

LED Indicators and Connectors

On the front side of the TA, there are three LEDs:

POWER

Lit The supply voltage has been applied.

PHONE

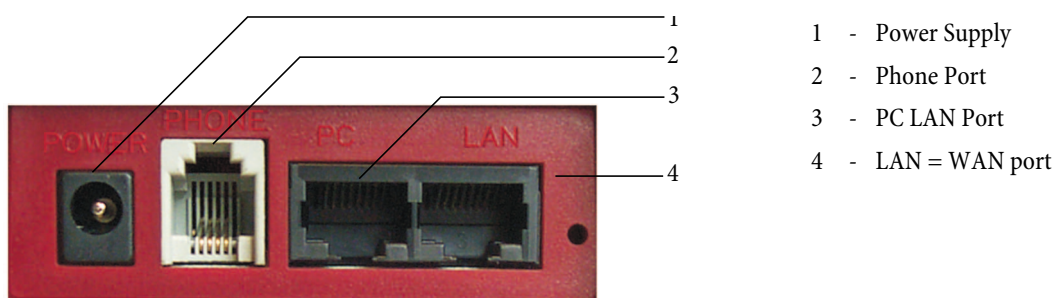
Lit The handset has been lifted up.

Flashes The TA has successfully registered with the SIP server.

LAN

Lit The Ethernet LAN port is physically connected with a peer device.

Please note the different labels of the WAN, LAN and of PC ports on the unit and on the PC configuration page.



Installation / SIP Configuration

- Use a Category 5 LAN cable to connect the »LAN« port of the TA to the ADSL modem or router.
- Use a Category 5 LAN cable to connect the »PC« port of the TA to the LAN port of the notebook or desktop PC.
- Connect an analog phone to the »PHONE« port of the TA. If the phone is fitted with a (German standard) TAE connector, insert the RJ11 to TAE adapter.
- Connect the power supply block. The »POWER« LED lights up.
- After approx. 5 seconds the »PHONE« LED flashes five times, then extinguishes as the unit goes to VoIP configuration mode.
- The »LAN« LED lights up when the »LAN port« is physically connected.
- Lift up the handset. The »PHONE« LED lights up and you hear the dial tone. If you hear the busy tone, please check whether the LAN port is correctly connected.
- On the phone dial #120# now to hear the TA's factory-preset IP address. The preset IP address is 192.168.123.001. This IP address will be used for the Web-based configuration from the PC. Starting on page 9 you will find additional details about the Web-based configuration.
- When the registration with the SIP server was successful (see page, Example 4) the »PHONE« LED flashes. Enter, for example, the subscriber number 123456 in order to call the party registered in the SIP server with the subscriber number 123456. Approx. 5 seconds after dialing the last digit of the subscriber number, you will hear the ringing tone. Alternatively you can press the # key after dialing the subscriber number to have dialing start immediately. When dialing the subscriber number without the # key, dialing will not start before the autodial timer expires (preset to 5 seconds).

V102 SIP VoIP Telephony Adapter

The V102 is a 2-port FXS telephony adapter (TA) with SIP protocol support for Voice-over-IP applications (VoIP). If the V102 is connected to the Internet and has two analog telephone sets attached, it supports two simultaneous VoIP calls. The V102 has Ethernet LAN and PC ports for interfacing with ADSL and a notebook PC. Additionally, it features two RJ11 connectors for analog telephone sets (FXS). The two FXS ports support T.38 functionality for telefax transmission through the Internet. Thanks to its integrated NAT/DHCP server the V102 is also easily configured for various network scenarios using a Web Browser on the PC or a telephone set, making it the ideal tool for providers (Internet Telephony Service-Providers) and SOHO users alike, who wish to make use of VoIP services.

Please note that the V102 requires one IP address, one subnet mask and one Gateway / Router IP address for connecting to the Internet. These three items of information will be supplied to you by your Internet Service Provider. The V102 can activate PPPoE or DHCP functions in order to have a dynamic IP address assigned automatically by the provider. Additional information can be found on page 9, (Configuration through the Web Browser).

V102 SIP VoIP Telephony Adapter Features

The V102 TA offers the following features:

- SIP v1 (RFC 2543), v2 (RFC 3261) with MD5 Authentication (RFC 2069 and RFC 2617).
- two RJ45 Ethernet ports and an RJ11 port for connection to analog terminal devices.
- ITU-T G.711, G.723, G.726, G.729A/B, VAD and CNG for Voice Codec.
- ITU-T G.165/168 Echo Compensation.
- Three LED Indicators: POWER, PHONE, LAN.
- Configuration supported through Web browser and telephone set.
- Integrated NAT/DHCP server.
- PPPoE/DHCP-Client für die dynamische Zuteilung der IP-Adresse, sowie NAT, DNS, und DDNS-Clients.
- Unterstützt STUN-Server für NAT-Traversal.
- Hot Line Mode support.
- Supports T.38 FAX via IP.
- Interactive voice response system (IVR) reads out IP status of phone.
- Remote Firmware Upgrade with HTTP or TFTP server through Web PC.
- Direkte IP/URL-Wahl ohne SIP-Proxy oder Rufnummer über SIP-Server.
- Telephony Features: Volume Adjustment, Phone Book, Speed Dialing, Call Forwarding, Call Waiting, Call Hold and Three-Party Conferencing, Redialing, and Flash.
- Outband DTMF (RFC 2833) / Inband DTMF / DTMF-SIP-Info senden.

Compatibility

The V102 VoIP TA supports the following standards:

VoIP Protocol:	IETF RFC 3261 and RFC 2543 for SIP.
SIP Authentication:	IETF RFC 2069 and RFC 2617 for MD5.
Voice codec:	see page 22, as well as VAD (Voice Activity Detection) and CNG.
Echo Compensation:	ITU-T G.165/168.

Package Contents

- V102 SIP VoIP Telephony Adapter.
- Power Supply Block 230V AC /12 VDC, 1 A .
- CD-ROM with User Manual.
- Brief description of functions.
- two RJ11 to TAE Adapters.

Bitte prüfen Sie, ob die Verpackung beschädigt ist oder ob ein Teil fehlt. If that is the case, please contact your dealer.

LED Indicators and Connectors

On the front side of the V102 there are three LEDs:

POWER

Lit: The supply voltage has been applied.

PHONE:

Solid green light: The handset of phone 1 or phone 2 has been lifted up.

Flashing green light: SIP registration for phone 1 was successful.

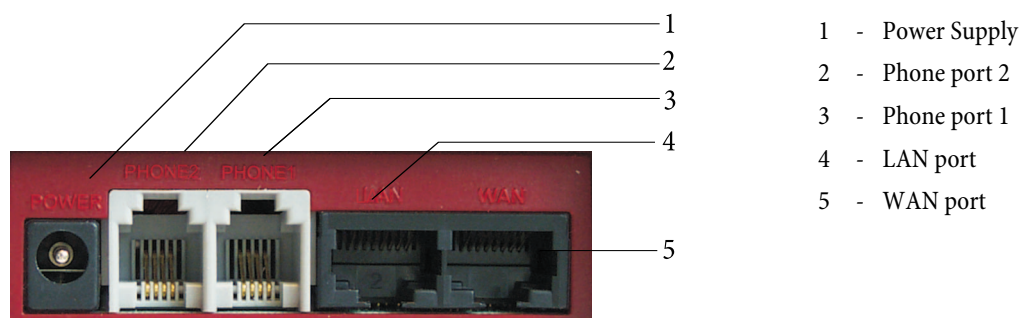
Flashing red light: SIP registration for phone 2 was successful.

Flashing yellow light: SIP registration for phone 1 and phone 2 was successful.

LAN

Lit: The Ethernet LAN port is physically connected with a peer device.

Flashing: Data traffic on Ethernet LAN ports.



Installation / SIP Configuration

- Use a Cat. 5 LAN cable to connect the »WAN« port of the TA to the ADSL modem or router.
- Use a Cat. 5 LAN cable to connect the »LAN« port of the TA to the LAN port of a notebook or desktop PC.
- Connect an analog phone to the »PHONE 1« port of the TA. If the phone is fitted with a (German standard) TAE connector, insert the RJ11 to TAE adapter.
- Connect a second analog phone to the »PHONE2« port of the TA.
- Connect the power supply block. The »POWER« LED lights up.
- After approx. 5 seconds the »PHONE« LED flashes five times, then extinguishes as the unit goes to VoIP configuration mode. The LAN LED is permanently illuminated when the »WAN port« is connected.
- Lift up the handset on phone 1 or 2. The »PHONE« LED lights up, and you hear the dial tone. If you hear the busy tone, please check whether the WAN port is correctly connected.
- On the phone dial #120# now in order to hear the TA's factory-preset IP address read out loud to you. The preset IP address is 192.168.123.001. This is the IP address you must enter in the Web browser in order to load the Web-based configuration page. Please refer to page for further details about Web-based configuration.
- On page 30 you will find examples for VoIP applications involving SIP registration as well as for registering phones 1 and 2 with your SIP server.
- The PHONE LED flashes in green when phone 1 has successfully registered or in red when phone 2 has successfully registered (yellow flashing indicates that both phones 1 and 2 have been registered successfully).
- After successful registration with the SIP server (see page Example 4) the »PHONE« LED flashes. Enter, for example, the subscriber number 123456 in order to call the party registered in the SIP server with the subscriber number 123456. Approx. 5 seconds after dialing the last digit of the subscriber number, you will hear the ringing tone. Alternatively you can press the # key after dialing the subscriber number to have dialing start immediately. When dialing the subscriber number without the # key, dialing will not start before the autodial timer expires (preset to 5 seconds).











Using Attached Analog Terminal Equipment

Not all of the service features described in the following are supported by all providers. If in doubt, consult with your provider to find out which service features are included in your access service package and which ones need to be ordered separately.

Symbols

Please note: Different terminal devices will not necessarily have the same dial tones, ringing signals or procedures for use. For further information on tone frequency and length, please refer to the assembly instructions.

Symbol Reference

	Lift up the handset of your telephone.
	This symbol indicates the call status. You have lifted up the handset of your telephone.
	This symbol prompts you to hang up or indicates that the telephone is idle.
	This symbol indicates some kind of signaling activity at your terminal, for example, that your phone is ringing.
	This symbol prompts you to dial the desired number.
	One of these symbols indicates that you should dial the digit shown, or a specific character.
	These symbols indicate a selection of digits or characters from which you can choose the appropriate one.
	This symbol prompts you to press the flash key (signal key).
	This symbol indicates that an acknowledgement signal can be heard in the handset.
	This symbol indicates that a conference call is in progress.

Telephony

Some terminal devices use their own special procedures for the functions described in the following. Refer to the operating instructions for the terminal device being used for detailed information.






When subscribers are put on hold they hear waiting music, depending on the provider.

Making External Calls




				
Lift up handset	Dial number	Wahl mit der Raute beenden	Conduct your call	Terminate your call

A call between two internal analog extensions is an outgoing local call if the connection is made through your provider.

Speed Dialing from the Phone Book

				
Lift up handset	Dial the desired speed dial destination (0...9)	Terminate dialing by pressing the # key.	Subscriber is being called	Conversation

Accepting Calls



			
Telephone is ringing	Lift up handset	Conversation	Terminate your call

Hot Line Calling

Making telephone calls with Direct Calling activated






	
Lift up handset	Begin dialing within the configured time

Room Inquiry




	
You are conducting a call. You would like to call another party	Press the Recall flash button. You hear the dial tone from the earpiece. The other party hears waiting music. They cannot hear you.

Inquiry Call

You can call another party at any time during an ongoing call.

		 	
You are conducting a call. You would like to call another party	Press the Recall flash button. You hear the dial tone from the earpiece.	Dial the number of the party you wish to call. Speak to the desired party.	If you replace the handset, you will receive a recall from the original caller.

Accepting a Waiting Call (Inquiry Call to Waiting Party)

		
You are conducting a call. A second party calls you. You hear the call waiting signal.	Press the Recall flash button.	You will be connected with the waiting party now. Your first call is put on hold (see »Inquiry call«). For further procedures refer also to »Broker's call«.






Enable Call Waiting


		
Lift up handset	Dial this code	

Disable Call Waiting

		
Lift up handset	Dial this code	

Broker's Call

		 	
You are in a call but wish to speak to a third party.	Press the Recall flash button.	Speak to that party. The original call is put on hold	You are conducting the call. The second call is put on hold.

 Press the R key in order to return to the original call, putting the second call on hold. By pressing the R key you can switch back and forth between both parties.

If you switch back and forth between two external parties (broker's call), then hang up your handset, the current call is terminated and the caller on hold will call you by way of a »Recall«.

Three-Party Conferencing

			
You are conducting a call with one party and wish to include a third party in the call.	Press the Recall flash button.	Dial the number of the party you wish to call.	You would like to include this party in the first call.

 Press the R key in order to return to the original call, putting the second call on hold. By pressing the R key you can switch back and forth between both parties.



The conference is ended when you hang up your handset. A party included in the conference call can hang up at any time and the party who initiated the call can then continue the conference with the remaining callers.

If you are in a conference call and you hang up, your call is terminated and the two parties from your conference remain connected with each other!

All settings for analog phones apply to both phones alike (exception: TA V101 »Service Domain Settings«).

Resetting the TA

The TA can be reset through any attached phone. Resetting deletes any data except for the content of the Phone Book. The entries from the Phone Book are removed only when resetting is done through the Web Configuration page. (»Default Settings«).

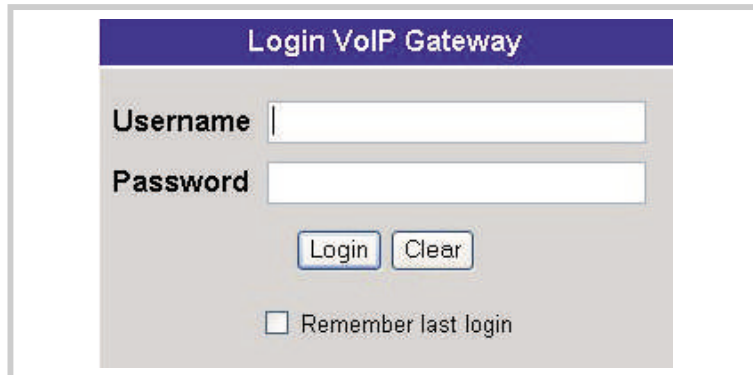


The TA will revert to the factory preset configuration and go to »POWER-ON« cycle. After four seconds the »PHONE« LED flashes five times, then extinguishes. The »POWER« LED lights up.

Configuration through the Web Browser (V101, V102)

You can configure the TA with a Web browser. To get started, first enter the IP address on the Web page.

- The factory-preset IP address is `http://192.168.123.1`.
- The following Web page will appear on your PC.



Enter the User Name and Password in the empty text boxes provided for that purpose. The preset values are:

- User Name: **root**
- Password: **test**

Click the «Login» button to go to the Management Information page for the system setup.

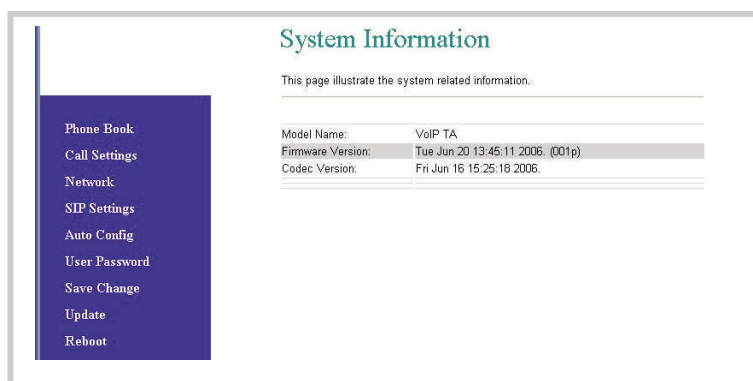
Please note: Whenever you change a setup on a Web page, be sure to click the page's »Submit« button followed by the »Save« button in order to have the new setup saved to non-volatile memory and activated.

System Information

The »System Information« page displays technical information like »Model Name«, »Firmware Version« and »Codec Version«.

Under »Model Name« you will find the name of the TA. In this field, »VoIP TA« identifies the V101 SIP VoIP telephony adapter and »VoIP FXS/FXS TA« is for the V102 SIP VoIP telephony adapter.

- Click the desired menu item at the left border in order to configure the TA.

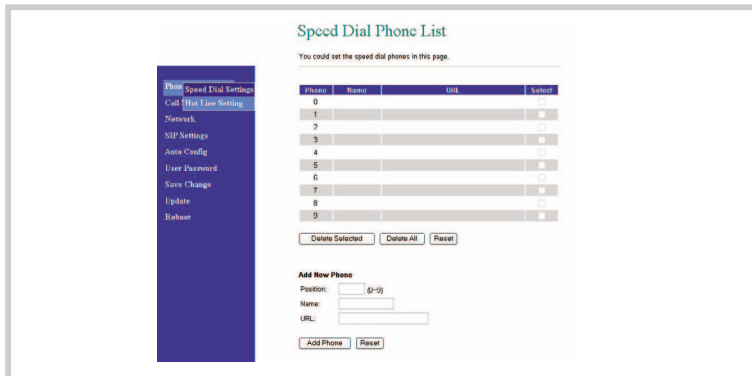


System Information	
This page illustrate the system related information.	
Model Name:	VoIP TA
Firmware Version:	Tue Jun 20 13:45:11 2006. (001p)
Codec Version:	Fri Jun 16 15:25:18 2006.

Phone Book

Phone Book / Speed Dial Settings

The »Phone Book« menu item lets you set up the Speed Dial feature under »Speed Dial Settings« and the Direct Call feature under »Hot Line Settings«. You can add, change or delete speed dial numbers in the speed dial list (»Speed Dial Phone List«).



Entering a Speed Dial Destination

Under »Add New Phone« you can enter a new speed dial destination.

- Under »Position« you may choose one out of 10 speed dial destinations (0...9). The digits 0 through 9 each represent a subscriber number. The TA dials these destination numbers automatically whenever you enter the appropriate digit for the desired subscriber.
- Under »Name« you can enter a name for each speed dial destination.
- Under »URL« you enter either the phone number, the URL, or the IP address of the subscriber you wish to store under this speed dial destination.
- When you are done, click the »Add Phone« button.
- By clicking the »Reset« button you can delete the entries made under »Add New Phone«.

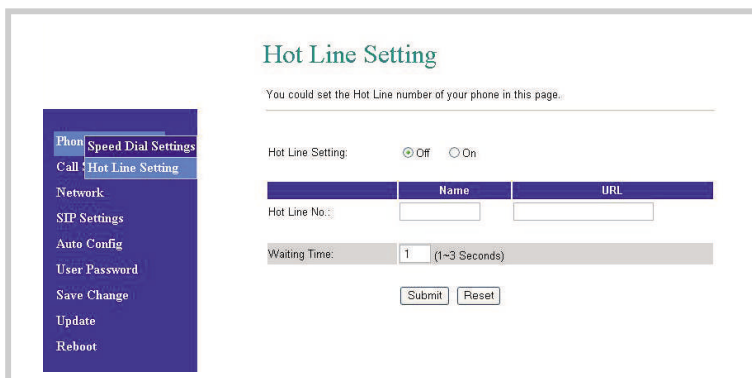
Once an entry has been added to the »Speed Dial Phone List« you cannot edit it anymore.

Removing a Speed Dial Destination

- Go to the »Select« column, choose the entries you want to remove, and click the »Delete Selected« button.
- If you wish to remove all subscriber numbers in one go, you need not choose all entries in the »Select« column. Simply click the »Delete All« button instead.

Direct Call (Hotline Setting)

If you lift up the handset of the phone but do not start dialing within the specified »Waiting Time« (factory preset to one second), a user-selected subscriber number will be dialed automatically.



Programming Direct Call Mode

- Under »Hot Line Setting« choose »On« in order to enable the Direct Call feature.

- Under »Hotline No.« enter the Direct Call destination.
- Under »Name« you can enter a name for the direct call destination.
- Under »URL« enter either the phone number, the phone number in URL format, or the IP address of the subscriber you wish to reach under the direct call number.
- Under »Waiting Time« enter the desired delay between the moment when you lift up the handset and the time when direct calling kicks in.
- Confirm your entries by clicking the »Submit« button.

Call Settings

The »Call Settings« menu item lets you access the following eight submenus:

- Call Forward
- SNTP
- Volume Settings
- Block Settings
- Caller Identification
- Autodial Timer (Dial Plan Setting)
- Flash Key (Flash Time Setting)
- Call Waiting
- T.38 FAX Setting

Call Forward

This feature lets you redirect your phone calls and incoming telefax messages to a location of your choice. For example, you wish to have a telefax message addressed to your office also sent to your fax machine at home.

Please note that not all providers support call forwarding. Before setting up this feature, you should therefore ask your provider for advice.

On the page »Forward Setting« you can select the forwarding mode, then enter the destination URL (phone number, phone number in URL format or IP address) to which incoming calls should be forwarded, as well as a name. The following call forwarding options are available:

Call forwarding on
No Answer
Forward):

An incoming call will be forwarded to the specified phone number, phone number in URL format, or IP address if the called subscriber does not pick up the handset within a specified time (as entered in »Time Out«).

Set Up Call Forwarding

You may enable all three call forwarding modes at the same time. In this case the TA will only use »Call Forward Unconditional« (All Forward). This call forwarding mode has the highest priority. The two remaining call forwarding modes may be active concurrently.

- To enable the desired call forwarding mode, choose »On«.
- Under »All Fwd No.:«, »Busy Fwd No.:« or »No Answer Fwd No.:« you can use the »Name« text box to enter a name for the forwarding destination.
- Under »URL« you enter either the phone number, the phone number in URL format or the IP address of the subscriber to whom calls should be forwarded.
- Under »No Answer Fwd Time Out« you specify the number of calls after which Call Forwarding on No Answer should kick in and redirect calls to the specified destination. You can choose any number between 2 and 8 ringing tones (»2~8 Ring«).
- Confirm your entries by clicking the »Submit« button.

Time Synchronization Setup (SNTP Settings)

To ensure that your TA is synchronized with standard network time, it is recommended to choose a time server. This server may be installed at an external location on the Internet as a so-called Public Time Server or reside in your own network. If you have configured a computer within your network as a Time Server, enter the IP address of that computer here.

SNTP

- To enable time synchronization choose »On«.

Time Synchronization

(Time Server)

- Under the »Primary Server« and »Secondary Server« setup items, enter the IP addresses of the »time synchronization sources«. You can enter two different addresses to ensure that time synchronization remains active off an alternative server when the primary server goes down.

Time Zone

(Time Zone)

- Here, enter the time difference between the standard time »Greenwich (Mean) Time« and your own location. For Central Europe the time zone needs to be set to + one hour (+ 01:00). Please note that Daylight Saving Time may be in force when setting the time zone.

Synchronization Time

(Sync. Time)

- Use this option to specify at which time interval (»Sync. Time«) resynchronization with the time server should take place.
- Confirm your entries by clicking the »Submit« button.

Volume Settings

- This option lets you set the volumes for voice communication. What you set here are the signal levels delivered to and received from the analog terminal.

Handset Volume

- Here you can enter the signal level that drives the volume in the speaker of the handset of your terminal.

Handset Gain

- Here you can enter the gain by which the volume received through the microphone of your phone set shall be raised.
- Confirm your entries by clicking the »Submit« button.

Please note that these settings are independent of any volume settings on your analog phone.

Block Settings

On this page you can set up the »Blocking« feature. Choose whether the phone should not ring at all (by setting »Always Block« to »On«) or whether ringing should be blocked for a certain period of time (»Block Period«).

No Ringing (Always Block)

Incoming calls will not ring your phone until you deactivate this function.

- To enable »Blocking« (Always Block) choose »On«.

Block Period

- The »SNTP Settings« must be properly set to be able to use this feature. To enable »Blocking« for a certain period of time, choose »On« for the »Block Period« option.
- In the »From« and »To« text boxes enter the start and end times of the period during which ringing should be suppressed. If the time of day specified for »From« occurs later than the one specified for »To«, ringing will be suppressed until that time on the following day.
- Confirm your entries by clicking the »Submit« button.

Please note that this feature requires an SNTP server.

Caller ID Setting

On the »Caller ID Setting« page you can set how callers are identified on the display of your phones.

Consult the User Manual of your phone which subscriber number signaling mode must be preset in the TA in order to accommodate the device.

Please note that the necessary settings are provider-specific.

Number Signaling Format

Depending on the type of terminal in use you must choose the appropriate format. Consult the User Manual of your phone to find out which number signaling format it supports.

You can choose between the following number signaling formats:

Don't show caller ID

- If your attached analog phone does not support caller identification or if you do not wish to have the caller's name and or number displayed, choose this menu option.

Caller ID after 1st Ring (FSK)

This setup is currently applicable for the following countries:

- Germany.

Caller ID before 1st Ring (FSK)

This setup is currently applicable for the following countries:

-

Caller ID before 1st Ring (DTMF)

For example, this setup is currently applicable for the following countries:

- Denmark.

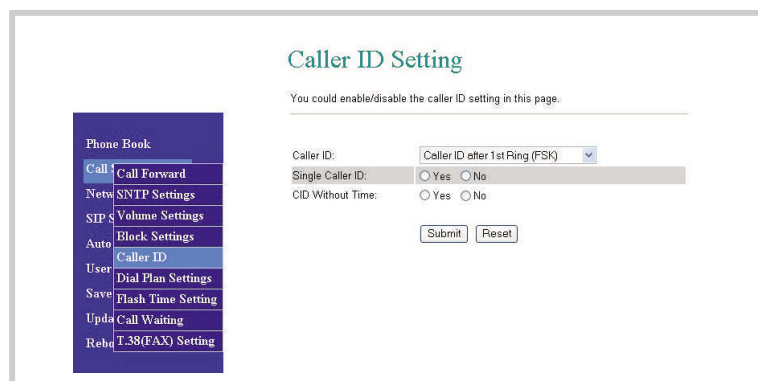
Do Not Show Name (Single Caller ID)

If your provider supports the identification of the calling party's number and name and your phone is able to understand that signaling information, you can view the name and number information on your display. Select »No« if this is desired. If you wish to have your phone display only the number but not the name, choose »Yes«.

CID Without Time

- If your provider does not support this function, choose »Yes«.
- Confirm your entries by clicking the »Submit« button.

These settings apply to both attached analog phones.



Dial Plan Setting

If you have enabled this service feature by choosing »ON«, the TA may recognize specific subscriber numbers or sequences of digits and process them in the following ways:

- Replace Rule
- Dial Plan
- Auto Prefix
- Prefix Unset Plan
- Auto Dial Time

Replacing Numbers (Replace rule)

By setting a »Replace Rule« you can have the TA replace a specific number you dial, for example, by a prefix and the same subscriber number. The replaced subscriber number may be 7 digits long at maximum.

Enter	Representing
x or X	0,1,2,3,4,5,6,7,8,9
+	or

Example:

Replace rule	001+006+009	->	005
--------------	-------------	----	-----

The sequence 001 or 006 or 009 is replaced with 005.

Dial Plan

The TA is able to detect a specific number and dial that number immediately without waiting for the # key that delimits the dialed sequence. Certain numbers, such as the emergency services number, are dialed immediately after the caller has entered the specific sequence of digits (example: emergency telephone number 999 used in the U.K.). The following entries are supported to enable the detection of numbers:

xx	Immediately dials specific two-digit function code numbers (for example PABX system control codes) if preceded by the asterisk () key.
#xx	Immediately dials specific two-digit function code numbers (for example PABX system control codes) if preceded by the number (#) key.
10x	Immediately dials specific telephone numbers (for example emergency service numbers) if the dialed number has three digits and starts with »10«.
11x	Immediately dials specific telephone numbers (for example emergency service numbers) if the dialed number has three digits and starts with »11«.
XXXXXxxx	Immediately dials specific telephone numbers (for example any number having the same digit count) if the dialed number has as many digits as specified in the »x« sequence.

Example

Speed dial	*xx+#xx+10x+11x+XXXXXXX
------------	-------------------------

Example explained:

Enter	Does What
*xx	Recognizes digits / sequences with: *00, *01, ..., *99
#xx	Recognizes digits / sequences with: #00, #01, ..., #99
10x	Recognizes sequences with: 100,101, ...,109
11x	Recognizes sequences with: 110,111, ...,119
XXXXXXX	Recognizes a-digit sequences

Auto Prefix

You may, for example, have the TA automatically place the country code or local area code before a telephone number you dial. For this to work, enter the desired telephone number under »Auto Prefix«.

Example:

Auto Prefix	0049	(00000000~99999999)
-------------	------	---------------------

Prefix Unset Plan

This function instructs the TA not to place the prefix specified under »Auto Prefix« before a subscriber number of your choice.

For this to work, the following conditions must be fulfilled:

- The TA must find a certain first digit in the number you dial.
- or
- The TA must find a certain number of digits in the number you dial.

Example:

Prefix Unset Plan	0+9+xxxx+xxxxxx
-------------------	-----------------

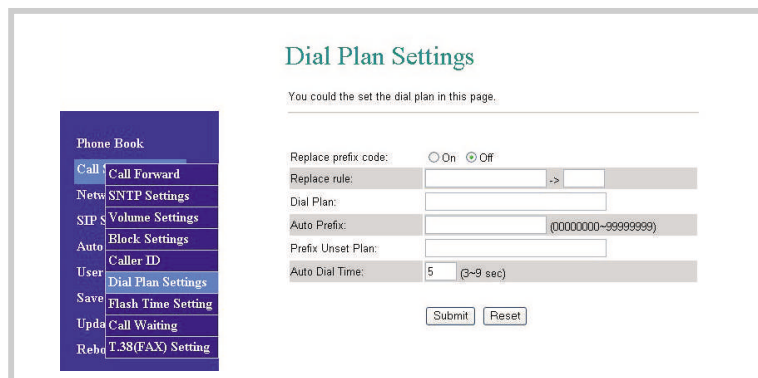
Example explained:

Enter	Does What
0	If the first digit is set to »0«, as in the example below, no prefix is placed before any number that starts with a »0«. This is the case if you dial a number with a prefix (for example 05171 1234567). This number is sent out exactly as you enter it.
+	This character means »or«. It allows you to combine several unset patterns, as shown in the example below.
9	If the first digit is a »9«, as shown in the example below, no prefix is placed before any number that starts with a »9«. This is the case if you dial a subscriber number like, for example, 9234567. This number is sent out exactly as you enter it.
+	Or
xxxx	Numbers made up of any combination of four digits are sent out exactly as entered, for example 1234.
+	Or
xXxXxX	Numbers made up of any combination of six digits are sent out exactly as entered, for example 123456.
	You can add more entries here, separated by a »+« character.

- Confirm your entries by clicking the »Submit« button.

Auto Dial Time

If you enter digits for composing a subscriber number, the TA must have some means of recognizing when dialing ends. During manual dialing, you mark out the end of a dialed number by pressing the number (#) key. If you do not press the number (#) key dialing is considered as completed when a certain period of time (»Auto Dial Time«) has elapsed after the last digit. You can adjust the length of that period of time between 3 and 9 seconds.



Flash Time Setting

The »Flash Time Setting« page lets you set the length of the Flash signal for the detection of the phone's Flash key. The Flash key is needed, for example, for Three-Party Conferencing and Call Hold. The detection period is factory-preset to a length of 600 milliseconds. The period can be adjusted to a length of up to 2550 milliseconds.

- Confirm your entries by clicking the »Submit« button.

Accepting a waiting call / Broker`s call (Call Waiting Settings)

The »Accept Waiting Call« feature allows you to accept a second call the presence of which is indicated by the call waiting tone. When you hear the call waiting tone, press the Flash key on your phone to be connected with the new caller. The party from the first call will be put on hold. Press the Flash key one more time to return to the first subscriber you have put on hold previously.

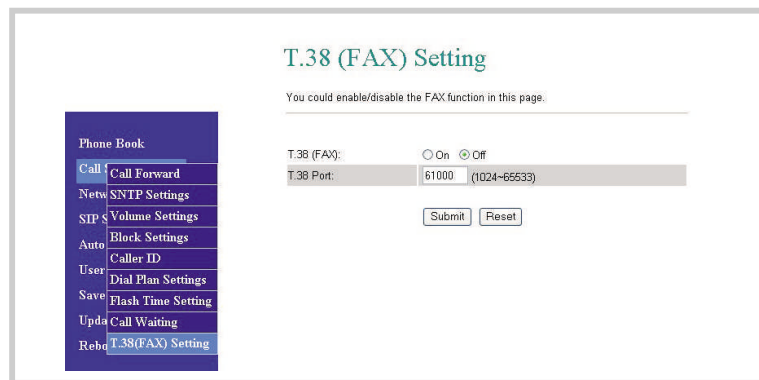
- If you wish to use this function, choose »On« under »Call waiting«.
- Confirm your entries by clicking the »Submit« button.

These settings apply to both attached analog phones.

T.38 (FAX) Setting

The T.38 FAX-over-IP feature enables fax machines to communicate over an IP network. In the factory default setting, neither port 1 nor 2 are set up for fax service (»Off«). By enabling this service you set up the TA to operate either with analog phones or analog fax machines attached to these ports.

- If you wish to use that feature, choose »On« under »T.38 (FAX)«.
- The two analog ports are already pre-configured. If fax machines attached to these ports do not work as expected, correct the port setup as instructed by your provider.
- Confirm your entries by clicking the »Submit« button.



Network

Network Status

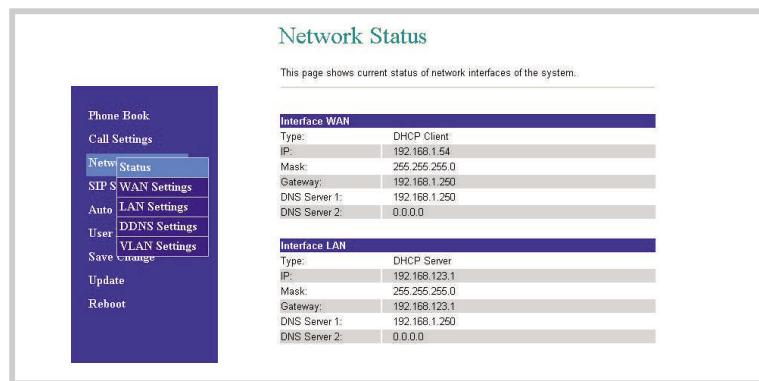
Difference between the LAN and WAN Ports of the V101 SIP VoIP Telephony Adapter

Please note that the labels of the RJ45 connectors of the TA and their equivalent options on the Configuration page does not match exactly!

- The »PC« port on the adapter is represented by the »Interface LAN« item on the Configuration page.

The »LAN« port on the adapter is represented by the »Interface WAN« item on the Configuration page.

By using the »Network Status« menu item you can check the current status of the WAN and LAN settings.



WAN Settings

This page lets you configure the network setup of the WAN port on the TA.

LAN Mode

- If LAN Mode is set to »Bridge« both Ethernet ports are physically interconnected in a fully transparent fashion. You need this option if the TA is interconnected with an external NAT router.
- If LAN Mode is set to »NAT«, the integrated NAT is activated. This setting is useful for ADSL users who don't have access to a NAT router.

IP Type

- **Fixed IP:**
Sets a fixed IP address for the WAN port.
- **DHCP client:**
Shows the IP address the TA has obtained from an upstream router or cable modem.

PPPoE

- Under «PPPoE Setting» you enter the »User Name« and »Password« given to you by your provider. In this mode the TA gets the IP address and the appropriate subnet mask from the DHCP server of your provider. Entries for the DNS server must be made manually. You may also enter a virtual MAC address if you have obtained one from your provider.

Confirm your entries by clicking the »Submit« button. The TA will reboot now.

LAN Settings

If you wish to set up the TA's integrated NAT on this page, you must deactivate Bridge mode beforehand.

LAN Settings

- The «LAN Settings» page lets you set up address translation for the network. Also featured on this page are options for setting the local IP address (IP), the subnet mask (Mask), and the MAC address (MAC).

DHCP Server

- These options enable (On) and disable (Off) the DHCP Server.
- Settings under »LAN Settings« are »Start IP«, »End IP«, the lifetime in days and hours of a dynamic IP address allocated by the DHCP server (»Lease Time«), as well as the activation status of the DHCP servers.

DynDNS Settings

DynDNS or Dynamic Domain Name System is a service on the Internet that maintains a fixed host name allocation for computer with a dynamically changing IP address. This services allows a PC to be addressed using an invariable domain name.

The »DDNS Settings« page lets you configure the DynDNS setup. Prior to entering the necessary information you need a DDNS account. Confirm your entries by clicking the »Submit« button.

VLAN Settings

A Virtual Local Area Network (VLAN) is a local virtual network that resides in a physical network. Its most widely used implementation is specified in the IEEE 8021Q standard.

There are two sections for configuring VLAN settings:

- one for VoIP packets that get through to the V101/V102,
- one for VLAN settings in NAT mode

Two kinds of packets may be received by the V101 /V102:

- VoIP packets
- packets that are forwarded to the PC through the LAN port

VLAN packets:

- If you activate the »VLAN packets« option, all incoming packets are verified for a correct IP address and VID, provided that you have made the appropriate settings for VID, User Priority and CFI.

VID:

- Please set the VID as specified by your Service Provider.

User Priority:

- defines a user priority using eight different priority levels (2^3). IEEE 802.1P specifies the use of the three user priority bits. Normally, this value is defined by your Service Provider.

CFI:

- The Canonical Format Indicator is always set to zero by Ethernet Switches. CFI is used to ensure the compatibility of Ethernet and Token Ring networks. When an Ethernet interface receives a frame with a CFI value of "One", that frame should not be forwarded to an unmarked interface without any processing.

If you activate the VLAN Packets option and enter values for VID, User Priority and CFI, all incoming packets that bear the IP address of the TA and the same VID will be accepted. Packets that have the IP address of the TA but a different VID are rejected by the TA. Packets with a different IP address are forwarded to the PC through the LAN interface.

NAT VLAN Settings:

If the V101 / V102 runs in integrated NAT mode, the TA filters and removes incorrect packets. Other devices downstream from the integrated NAT may be subdivided in 4 VLAN groups by assigning these groups different VIDs. The V101 / V102 verifies the VID in the incoming packets from the ADSL modem. When a packet arrives with an IP address other than that of the TA but with the correct VID (or without any of the four user-defined VIDs), the TA will reject it.

SIP Settings

This menu item lets you set the Service Domain, make Codec settings, Codec ID settings and miscellaneous other settings.

You can make calls to and receive calls from other parties through the SIP account.

Phone No. This option is only available with the TA 102. Here, you can select the telephone that you want to have registered.

Active: Click «ON» to set the Service Domain to «Active» status, then enter the following data:

Display Name: Enter the name that you wish to have displayed.

User Name: Enter the User Name given to by your provider.

Register Name: Enter the register name specified by your provider.

Register Password: Enter the register password specified by your provider.

Domain Server: Enter the Domain Server specified by your provider.

Proxy Server: Enter the Proxy Server specified by your provider.

Outbound Proxy: Enter the Outbound Proxy address used by your provider. If unknown, you may skip his option.

- If the »Status« row reads »Registered«, this indicates successful registration with the provider. The »PHONE« LED starts flashing in that case. The TA is now ready for VoIP calls.
- If you have more than one SIP account, please carry out all of the steps described above in order to register with the alternative providers.
- Confirm your entries by clicking the »Submit« button.

DTMF Setting

Under »DTMF Setting«, you can specify options for DTMF dialing. The options include RFC 2833 (Outband DTMF), Inband DTMF, and DTMF SIP Info. The factory default setting is Outband DTMF »2833«. To use DTMF in-call dialing (for example 2833), you may have to select the Inband DTMF option. You may need DTMF in-call dialing when you call, for example, a public agency and your calls are handled by an automated interactive voice recognition system. Once you are connected the voice recognition system may prompt you to enter a certain sequence of digits. By doing so, you may be automatically put through to a department that can answer your inquiry.

Port Setting

Under »Port Setting« you can specify the SIP and the RTP port numbers. Providers may be using different SIP/RTP port settings. Please consult with your provider in order to learn about the correct port numbers to be entered here. Confirm your entries by clicking the »Submit« button.

Factory Default Setting: »Phone 1« SIP port 5060, RTP port 60000.

»Phone 2« SIP port 5061, RTP port 60100.

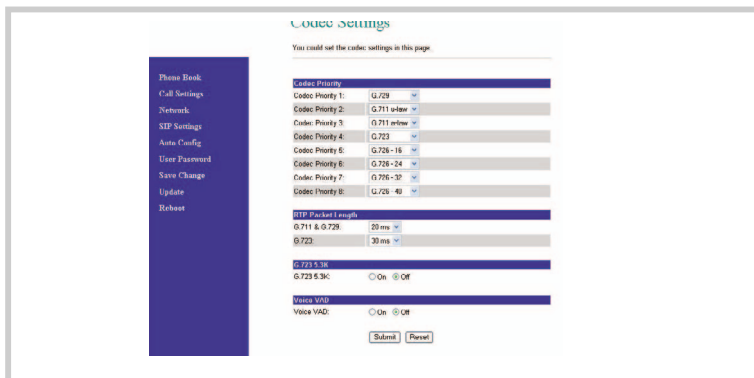
STUN Setting

Under »STUN Setting« you can set up the STUN server for the SIP service. The STUN function must be activated in order for a TA registered with the SIP server to work correctly downstream from a NAT server. You can enter the IP address of the STUN server and the STUN port number.



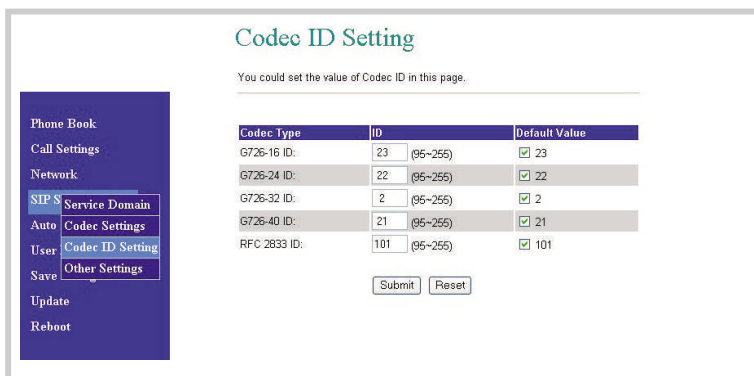
Codec Setting

The »Codec Settings« page lets you set »Codec Priority«, »RTP packet length« and configure the VAD function. Be sure to enter this data exactly as specified by your provider.



Codec ID Setting

On the »Codec ID Setting« page, you can specify the Codec ID. Beim Eingeben dieser Daten müssen Sie die Vorgaben des Provider befolgen.



Other Settings

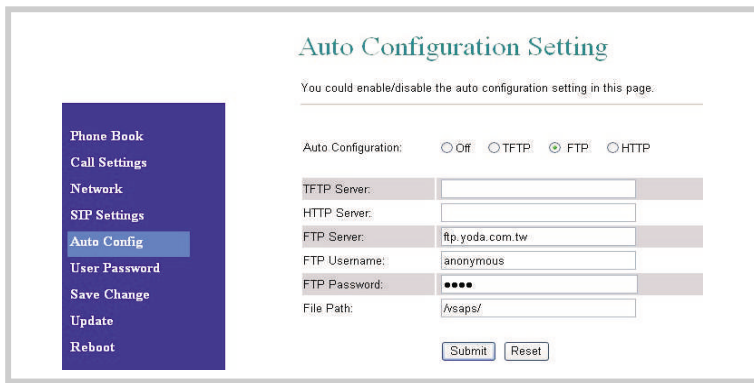
On the »Other Settings« page you can configure Hold mode through RFC and QoS (Quality of Service). Please follow the instructions given by your provider when changing these settings.

Confirm your entries by clicking the »Submit« button.

The QoS value allows for prioritizing voice packets. Higher values mean higher priority for voice packets on the Internet. It should be noted that any intervening systems on the Internet must support the QoS function.

Automatic Configuration Setup (Auto Config)

The »Auto Config« menu item takes you to the »Auto Configuration Setting« page. This page allows you to download configurations stored on a TFTP or FTP server using the auto configuration function. This setup approach may come in handy for new users, allowing them to get started right away by automatically downloading a ready-for-use configuration setup. Before you can activate Auto Configuration, you must go to the »Network« setup page and select the »Bridge« and »Fixed IP« modes. After activating that function by clicking »On«, click the »Submit« button. Do not forget to click the »Save« button in the »Save Change« window. The TA reboots now, then downloads the predefined setup automatically from the TFTP or FTP server.



User Password

The »User Password« menu item takes you to the »User Password« page. On that page, you can change your »Username« and »Password«.

New username

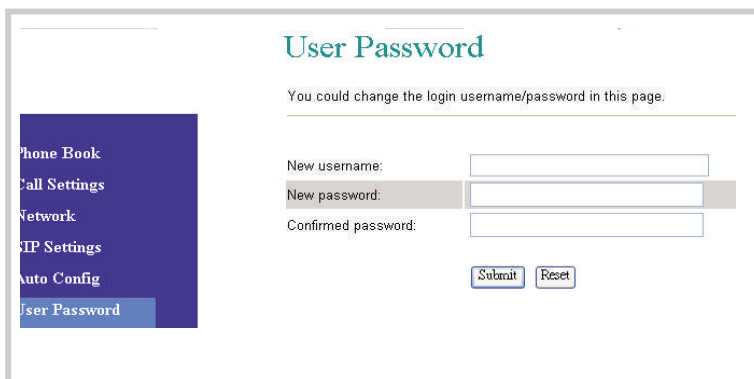
- Use this text box to enter your new User Name.

New password

- Use this text box to enter a new password.

Confirmed password

- Use this text box to enter your new password one more time.
- Confirm your entries by clicking the »Submit« button.



Save Changes

On the »Save Changes« page you can permanently save the changes you made to the TA setup.

- Click the »Save« button.

After clicking the »Save« button the TA reboots automatically. The new settings become effective when booting is complete.



Update

With the »Update« menu item and the »New Firmware« submenu item the TA offers two methods for loading the latest firmware in a HTTP or TFTP session.

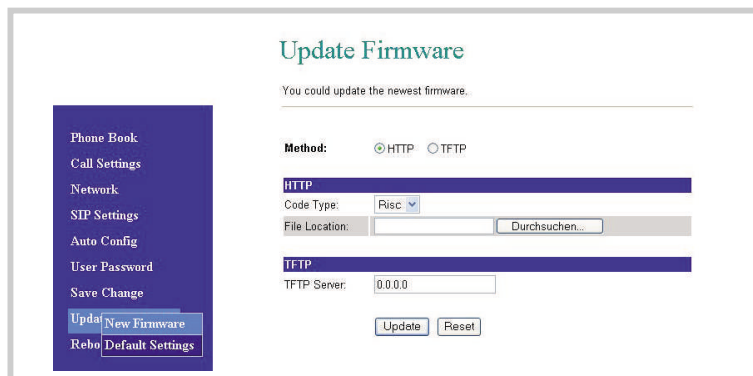
- Select the desired type of server, HTTP or TFTP.

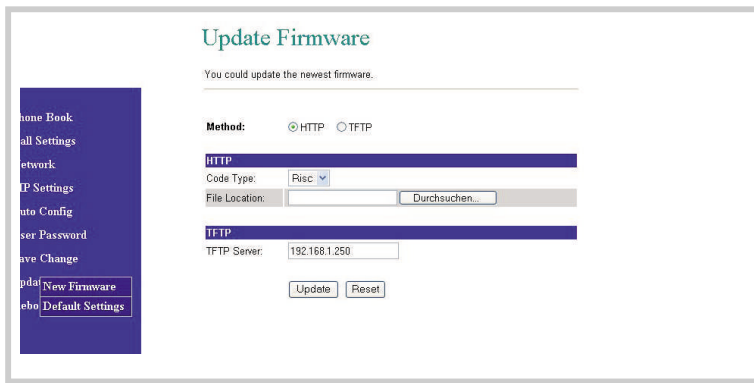
HTTP

- Now select the type of firmware code (»Code Type«): Risc or DSP code (Risc code is the preferable option).
- **Code Type:**
Click the »Browse« button in order to select the storage location for HTTP.
- When you are done, click the »Update« button.

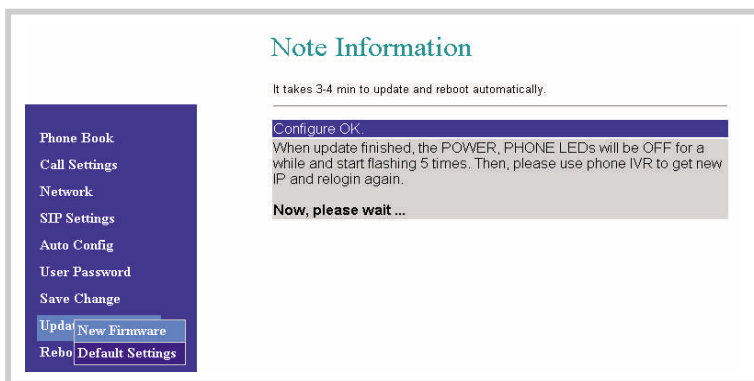
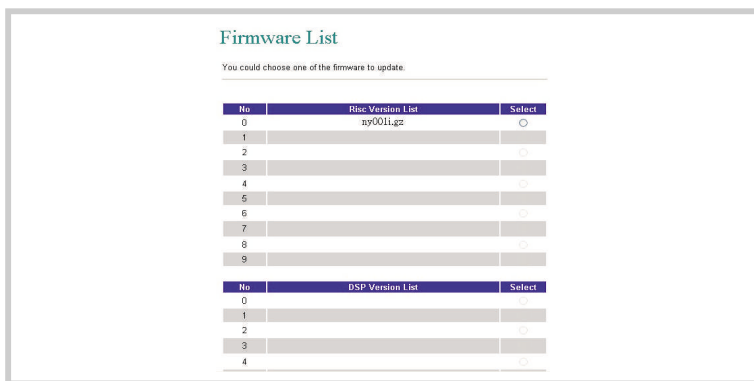
TFTP

- Enter the IP address of the TFTP server from which you wish to download firmware.
- When you are done, click the »Update« button.





- After clicking the »Update« button, you will be presented with a list of firmware items posted on the server for your to download.
- Select the new file you wish to download to your TA, then click the »Select« button.
- Confirm your entries by clicking the »Submit« button.



- After 3 to 4 minutes the »PHONE« LED flashes five times, thus indicating the successful completion of the firmware update. Now log back in and dial #120# to have the new IP address for the Web-based configuration identified.

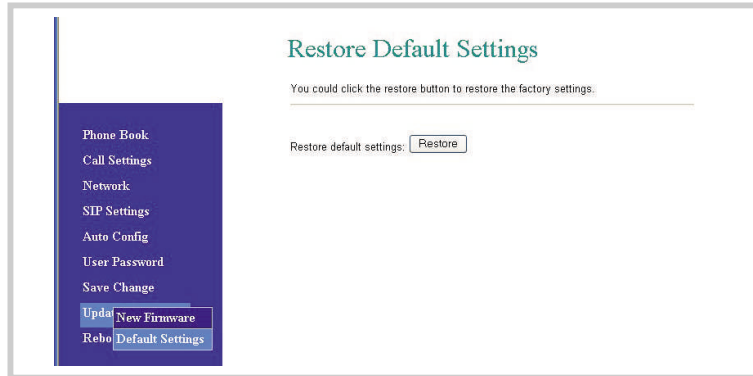
Be sure not to deactivate the TA after clicking the »Select« button. Doing so may damage the unit.

Downloading from a TFTP server works only with public IP addresses.

Default Settings

The »Default Settings« submenu item takes you to the »Restore Default Settings« page. On that page, you can reset the TA to its factory default settings.

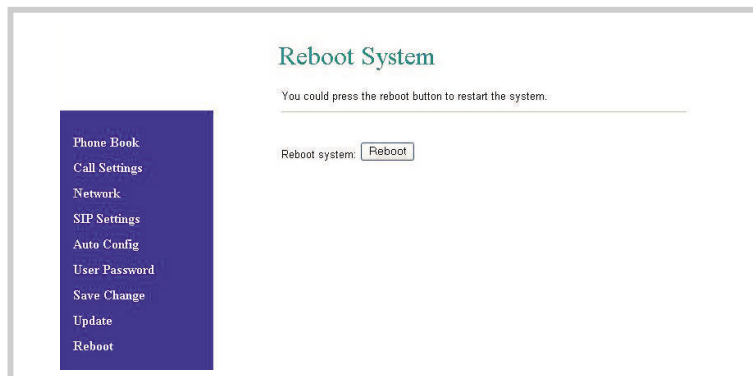
- When you click the »Restore« button the TA returns to its factory default settings and automatically initiates a system reboot.



Reboot

The »Reboot« menu item takes you to the »Reboot System« page. On that page, you can initiate a reboot of the TA.

- Click on the »Reboot« button to have the TA reboot automatically with the previously selected configuration.



Configuration using the Telephone's Keypad

You can configure the TA from the keypad of your phone and check its status. For this to work, the LAN port must be connected to the Ethernet, otherwise you hear the busy tone from your earpiece.

The voice output is given in English only.

	Status Check	
Interactive Voice Response System (IVR) Action	Entry via Telephone Keypad	Remarks
Query for DHCP Type	#121#	The voice response system reads out whether DHCP is activated (»DHCP«) or deactivated (»Fixed IP«). Hang up when you hear the end of announcement signal.
Check LAN (V102) or PC (V101) port IP address of the TA	#120#	The voice response system reads out the current IP address of the TA. Hang up when you hear the end of announcement signal.
Check subnet mask	#123#	The voice response system reads out the current subnet mask of the TA. Hang up when you hear the end of announcement signal.
Check router IP address	#124#	The voice response system reads out the current IP address of the router. Hang up when you hear the end of announcement signal.
Check setup of primary DNS server	#125#	The voice response system reads out the current value from the primary DNS field.
Check WAN (V102) or LAN (V101) port IP address of the TA	#126#	The voice response system reads out the current WAN IP address of the TA. Hang up when you hear the end of announcement signal.
Check firmware version	#128#	The voice response system reads out the firmware version.

	Settings	
DHCP Client Setup	#111#	This setup activates the DHCP client.
Setup of Static IP Address for the TA	#112xxx*xxx*xxx*xxx#	Example: #112061*066*159*009# Please note: xxx must be substituted with 3 decimal numbers. This setup deactivates the DHCP client.
Setup of Subnet Mask	#113xxx*xxx*xxx*xxx#	Example: #113255*255*255*000# Please note: xxx must be substituted with 3 decimal numbers. The static IP address of the TA must be set first (#112).
Setup of Router IP Address	#114xxx*xxx*xxx*xxx#	Example: #114061*066*159*254# Please note: xxx must be substituted with 3 decimal numbers. The static IP address of the TA must be set first (#112).
Setup of Primary DNS Server	#115xxx*xxx*xxx*xxx#	Example: #115159*168*001*001# Please note: xxx must be substituted with 3 decimal numbers. The static IP address of the TA must be set first (#112).

Setup of Codec	#130+[1-8]#	1: G.711 u-law, 2: G.711 a-law, 3: G.723.1,4: G.729a, 5: G.726-16K,6: G.726-24K, 7: G.726-32K,8: G.726-40K.
Setting Microphone Gain	#131+[00~15]#	Example: #13107#, factory default setting is 10.
Setting the Handset Volume	#132+[00~12]#	Example: #13209#, factory default setting is 10.
Activating Call Waiting	#138#	
Deactivating Call Waiting	#139#	
Rebooting	#195#	Hang up. The TA reboots automatically.
Reset to the factory default setting	#198#	The TA will be returned to the factory default settings. WARNING: ALL diverging, user-edited settings will be lost!

Application Examples

Example

ADSL link without NAT router serving the TA

Application:

This is a simple application scenario downstream from an ADSL modem, which is normally supplied to you by your provider.

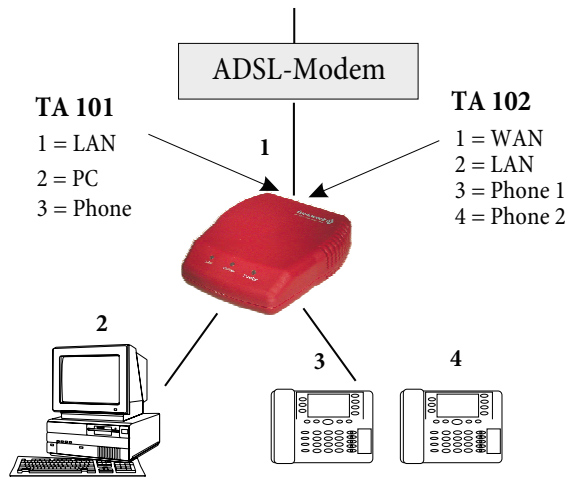


Bild 1

- Connect your TA as shown in the illustration 1 .
- Set up your PC for DHCP mode, so it obtains its IP address from the TA.
- Enter <http://192.168.123.1> to access the Web-based configuration page.
- Choose »Network«, then the »WAN Settings« page
- Set the LAN mode to »NAT«.
- Under »IP Type«, select »PPPoE«.
- Under »WAN Settings«, »PPPoE Setting«, enter the User Name and Password specified by your provider.
- Confirm your entries by clicking the »Submit« button. The TA will reboot now.
- In the web browser on your PC, enter the desired web address.

Example2

ADSL links through ADSL cable modem or a NAT router serving the TA

In DHCP mode:

- The IP address of the TA will be obtained from the cable modem or router.

Fixed IP address:

- The IP address of the TA is equivalent to an IP address of the cable modem or an IP address from the router's network.

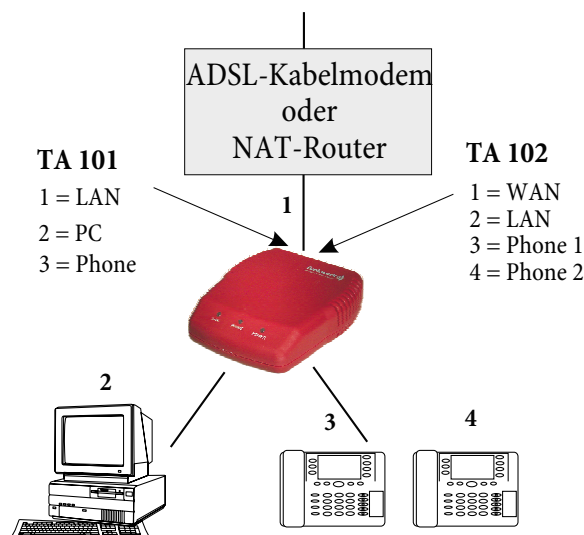


Bild 2

Configuration:

- Connect your TA as shown in the illustration 2.
- Set up your PC for DHCP mode, so it automatically gets its IP address from the TA.
- Enter <http://192.168.123.1> to access the Web-based configuration page of the TA.
- Choose »Network«, then the »WAN Settings« page
- Set the LAN mode to »NAT«.
- Under »IP Type«, select »DHCP Client« to obtain the IP address automatically from the DHCP-server.
- Confirm your entries by clicking the »Submit« button. The TA will reboot now.
- Geben Sie im Browser Ihres PCs die gewünschte Internetadresse ein.

A TA attached to a router becomes a part of the router's network, thus allowing it to be accessed and configured by entering its respective IP address on a PC that also resides on the network managed by the same router.

Example3

Direct IP Call

Application:

The following applications are supported for ADSL links without NAT routers. Both subscribers must have fixed or dynamic IP addresses. For the direct IP call to work both the calling as well as the called party must have well-known, fixed or dynamic IP addresses.

Configuration:

- Choose the «Network « menu item, then » WAN Settings«
- Set the LAN mode to »NAT«
- Set the »IP Type« option to »Fixed IP«.
- Enter the »IP«, »Mask«, and »Gateway« addresses.

- Confirm your entries by clicking the »Submit« button.
- Check whether the SIP server is deactivated (Note: Off is default) and that the »PHONE« LED is not flashing.

Making a call:

- Pick up the handset of your analog phone.
You hear the dial tone.
- Dial **211*21*191*4#**
or **211*021*191*004**

to call the party with the “real” IP address 211.21.191.4. Please note: If you enter a subscriber number followed by »#«, that subscriber will be dialed up right away. When you dial the subscriber number without the # key, dialing will not start before the autodial timer expires (preset to 5 seconds). You will hear the ringing signal while the other party is being rung.

Example4

Point-to-Point SIP Calls

Application:

This example shows ADSL links involving a NAT router. Both subscribers are registered under a NAT router in the SIP server either with their fixed “real” IP address or with a private IP address. For point-to-point SIP calls to work, both the calling as well as the called parties need to be registered with the SIP server with their specified registered phone numbers.

Configuration:

- Choose the menu item »Network«, then »WAN Settings«.
- Set the LAN mode to »Bridge«.
- Set »IP Type« to »Fixed IP«.
- Enter the »IP«, »Mask«, and »Gateway« addresses.
- Confirm your entries by clicking the »Submit« button.
- Choose »SIP Settings«, then »Service Domain«.
- Under »Realm« set »Active« to »On«.
- Enter the data for »Register Name«, »Register Password«, »Proxy Server« and »Outbound Proxy«.
- Under »STUN Setting«, set »STUN« to »On« when no Outbound Proxy is available.
- Confirm your entries by clicking the »Submit« button.
- After successful registration the »PHONE« LED starts flashing.

Making a call:

- Pick up the handset of your analog phone.
You will hear the dial tone now.
- Now dial, for example, **12345** in order to call the desired subscriber with the number 12345.
- You will hear the ringing tone, while the desired subscriber is being rung.

Example5

SIP to Direct IP Calling

Applications:

This example shows subscribers with ADSL links, as illustrated in Figure 1 or figure 2. The calling subscriber is registered in the SIP server either with his or her fixed / dynamic IP address or a private IP address from the NAT router. The called subscriber has a fixed “real” IP address.

This example assumes that the router upstream of the TA supports that scenario. It should also be noted that the provider may offer direct calling via the IP address.

Configuration

- The configuration is equivalent to example 4.

- On the »Service Domain Setting« page, under »STUN Setting«, set »STUN« to »On« if there is NO Outbound Proxy available.
- Observe whether the »PHONE« LED flashes, thus indicating that the registration with the SIP server was successful.

Making a call:

- Pick up the handset of your analog phone.
You hear the dial tone.
- Dial **2 1 1 * 2 1 * 1 9 1 * 4 #**
 - o **2 1 1 * 2 1 * 1 9 1 * 4**

to call the subscriber with the fixed IP address 211.21.191.4.
- You will hear the ringing tone, while the desired subscriber is being rung.

Example 6

Making Calls through a SIP-Provider Using the TA

Applications:

In this example, we will use the German provider QSC as an example to demonstrate how you can set up the TA to use an independent provider. The application involves an ADSL access with a NAT router. Both subscribers must be registered with QSC's SIP server.

- Go to the Web page <http://www.qsc.de> and sign up there in order to obtain a registration number.
- After successful registration, QSC provides you with a QSC subscriber number and a password. The registered name or the registered subscriber number may, for example, be 0211546088, while the password may be xxxx.
- Log on to the Web-based configuration page.

Configuration

WAN Settings

- Choose the menu item »Network «, then »WAN Settings«.
- Set the LAN mode to »NAT«.
- Set »IP Type« to »DHCP Client« if you wish to obtain an IP address automatically from your provider.
- or
- Set »IP Type« to »PPPoE« and enter the User Name and Password specified by your provider.
- Confirm your entries by clicking the »Submit« button.

The screenshot shows the 'WAN Settings' configuration page. On the left is a dark blue navigation menu with white text: 'Phone Book', 'Call Settings', 'Network', 'SIP Settings', 'Auto Config', 'User Password', 'Save Change', 'Update', and 'Reboot'. The main content area has a title 'WAN Settings' and a subtitle 'You could configure the WAN settings in this page.' Below this, there are several configuration options:

- LAN Mode:** Radio buttons for 'Bridge' and 'NAT' (selected).
- WAN Setting:** A sub-section header.
- IP Type:** Radio buttons for 'Fixed IP', 'DHCP Client' (selected), and 'PPPoE'.
- IP:** Text input field containing '192.168.30.112'.
- Mask:** Text input field containing '255.255.255.0'.
- Gateway:** Text input field containing '192.168.30.254'.
- DNS Server1:** Text input field containing '192.168.30.254'.
- DNS Server2:** Text input field containing '0.0.0.0'.
- MAC:** Text input field containing '00a0f00b-c01'.

LAN Settings

- Now choose the »Network « menu item, followed by »LAN Settings«.
- Under »DHCP Server«, set the »DHCP Server« option to »On« if you wish to obtain the IP-address from the DHCP-server.
- Confirm your entries by clicking the »Submit« button.

LAN Settings

You could configure the LAN settings in this page.

LAN Setting

IP:

Mask:

MAC:

DHCP Server

DHCP Server: On Off

Start IP:

End IP:

Lease Time: : (dd:hh)

SIP Settings

- On the »Service Domain Settings« page enter the User Name (including the Display »Name«, »User Name«, »Registered Name«), the registered password, the Domain Server (sip.qsc.de), the Proxy Server (sip.qsc.de) and the Outbound Proxy (sip.qsc.de). Confirm your entries by clicking the »Submit« button, then click the »Save Change« button. The system reboots automatically now. When the system has finished rebooting, the page with the SIP setup displays »Registered«, and the »PHONE« LED flashes.

Service Domain Settings

You could set information of service domains in this page.

Phone No.:

Realm

Active: On Off

Display Name:

User Name:

Register Name:

Register Password:

Domain Server:

Proxy Server:

Outbound Proxy:

Status: Registered

DTMF Setting

2833

Inband DTMF

Send DTMF SIP Info

Port Setting

SIP Port: (1024-65535)

RTP Port: (1024-65535)

STUN Setting

STUN: On Off

STUN Server:

STUN Port:

Registered Name in the QSC-SIP server: 02115426988 Password: xxxx

The screenshot shows the 'Codec Settings' page. On the left is a vertical navigation menu with the following items: Phone Book, Call Settings, Network, SIP Settings, Auto Config, User Password, Save Change, Update, and Reboot. The main content area is titled 'Codec Settings' and contains the following settings:

- Codec Priority 1:** G.729
- Codec Priority 2:** G.711 ulaw
- Codec Priority 3:** G.711 alaw
- Codec Priority 4:** G.723
- Codec Priority 5:** G.726 - 16
- Codec Priority 6:** G.726 - 24
- Codec Priority 7:** G.726 - 32
- Codec Priority 8:** G.726 - 40

Below the codec priorities are two sections:

- SIP Packet Length:**
 - G.711 & G.729: 20 ms
 - G.723: 30 ms
- Voice VAD:**
 - G.723 & 3G: On Off
 - Voice VAD: On Off

At the bottom of the form are 'Submit' and 'Reset' buttons.

Making a call

- Pick up the handset of your analog phone.
You hear the dial tone. (Your QSC-subscriber number is 02115426988).
- Now dial, for example **1 2 3 4 5 #** in order to call the subscriber with the registered subscriber number 12345. Please note: # dials the subscriber number without any delay. When you dial the subscriber number without the # key, the dialing will not start before the autodial timer expires (preset to 5 seconds). You will hear the ringing tone, while the desired subscriber is being rung.

Troubleshooting the Web-based Configuration

Access the Web page impossible

The Web browser on your PC is the tool for configuring the TA. If you enter the preset address `http://192.168.123.1` in the PC-based Web browser for the very first time (see Figure 5) but you cannot access the TA, the PC may be set to work with an IP-address other than 192.168.123.xxx. In this case you must change the IP address of the PC.

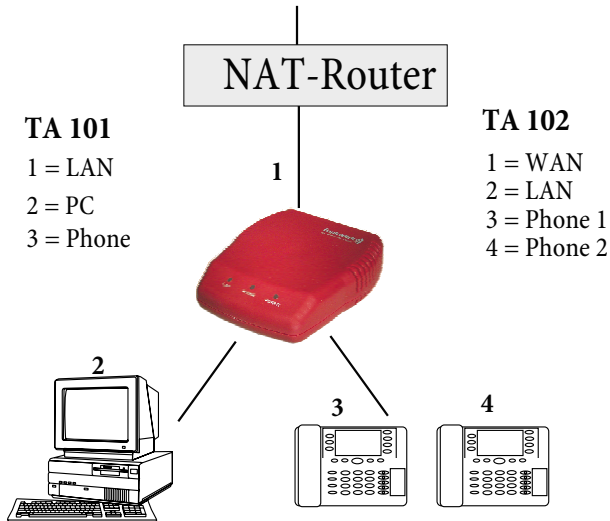


Bild 5

Example: Setting the IP address of the PC:

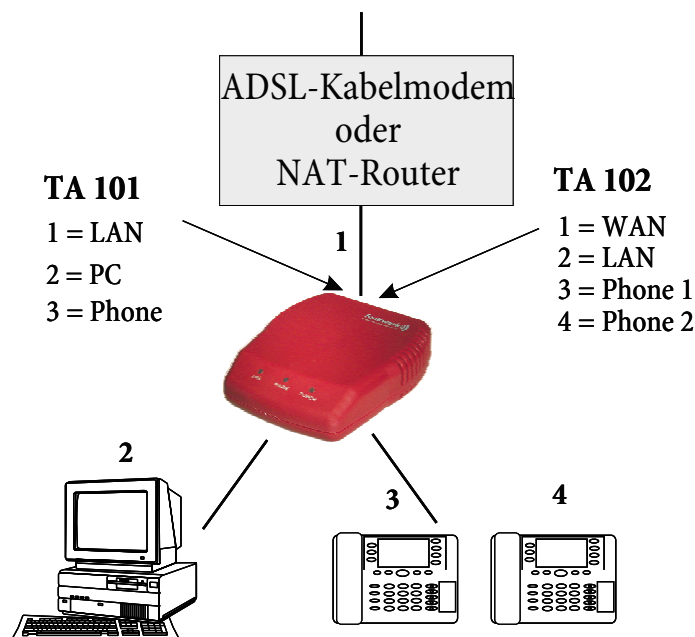
Windows 2000:

- Choose »Start«, »Settings«, »Control Panel«, »Network and Dial-up Connections«, then click »Local Area Connection«, »Properties«.
- The »Local Area Connection Properties« window opens now.
- Double-click »Internet Protocol (TCP/IP)«.
- Click »Use the following IP address«.
- Enter the IP address for the PC (2): 192.168.123.50. Instead of "50" you may enter any number between 002 and 254. The 001 is reserved for use by the TA.
- Enter the subnet mask: 255.255.255.0
- For the Standard Gateway enter (1): 192.168.123.1
- Click the »OK« button.
- In the Web browser enter `http://192.168.123.1`.
- Activate the DHCP server for LAN setup (2).
- Now, in the WAN setup (1) choose »DHCP Client« in order to coordinate with your ADSL / cable modem.
- Click »Save change« and wait until the TA has rebooted.

All connections will be automatically shut down now.

- On your PC set the »Internet Protocol (TCP/IP) Properties« option to »Obtain an IP address automatically«.
- In the Web browser enter `http://192.168.123.1`.
- Carry on with the SIP configuration for the TA.

Short Reference Manual



Broadband Link through the TA (ADSL/cable modem)

- Connect the LAN port of the TA to the ADSL NAT router.
- Connect the PC to the TA using a Category 5 LAN cable.
- Connect an analog phone to the respective »PHONE«, »PHONE 1« or »PHONE 2« port of the TA.
- Connect the power supply block to the TA. The »POWER« LED lights up. After 5 seconds the »PHONE« LED flashes five times. The TA is now ready for accepting configuration entries.
- Lift up the handset of your phone. The »PHONE« LED lights up, and you hear the dial tone.
- Now dial **# 1 2 1 #**. The voice response system built into in your TA reads out the DHCP status of the TA to you.
Hang up now.
- Now dial **# 1 2 0 #**. The voice response system built into your TA reads out the IP address now (for example 192.168.123.1).
Hang up now.
- Connect your TA as shown in the illustration below.
- Set up your PC for »DHCP« mode. Doing so makes the PC obtain its IP address from the TA.
- Enter the IP address <http://192.168.123.1> for accessing the Web-based configuration page.
- Choose »Network«, then go to the »WAN Settings« page.
- Set the LAN mode to »NAT«.
- Under »IP Type«, select »PPPoE«.
- Under »WAN Settings«, »PPPoE Setting« enter the User Name and Password specified by your provider.
- Confirm your entries by clicking the »Submit« button. The TA will reboot now.
- In the web browser on your PC, enter the desired web address.

ADSL links with external NAT router serving TA

- In order to be automatically assigned an IP address from a NAT router, the TA must be set up as a »DHCP Client«.
- From an attached analog phone dial, for example, **# 1 2 0 #** in order to have the IP address for the TA read out loud to you for verification.
- In the Web browser enter the IP address for the TA configuration page (factory default setting is 192.168.123.1).

Making Point-to-Point SIP Calls

- When the TA has registered with the SIP-Server, the »PHONE« LED flashes.
- Pick up the handset of your analog phone.
You will hear the dial tone.
- Enter a subscriber number, for example **1 2 3 4 5 6 #** in order to reach the subscriber who is registered in the SIP server with the subscriber number 123456.

Please note: When you enter **#** the call will be forwarded immediately. Shortly afterwards, you will hear the ringing tone from the called party.

Index

A

All Forward	11
Always Block.	13
Application Examples.	30
Auto Config	23
Auto Dial	15
Auto Dial Setting.	11,15
Auto Dial Time	15
Auto Prefix.	15
Autodial Timer	11
Automatic Configuration Setup	23
Automatic Dial Time	15
Automatic Prefix	15

B

Block Period	13
Block Settings	11,13
Blocking Auto Prefix	15

C

Call Forward	11
Call forwarding	11
Call Forwarding	11
Call Forwarding on No Answer	12
Call Settings	11
Call waiting.	17
Call Waiting	11,17
Call Waiting Setting.	17
Call Waiting Settings	17
Caller ID	11,14
Caller ID Display	14
Caller Identification.	11
Codec	22
Codec ID	22
Codec ID Setting.	22
Codec Settings	22
Compatibility	2,4
Configuration through the Telephone's Keypad	28
Configuration through the Web Browser	9
Confirmed password	24

D

DDNS	20
DDNS Settings	20
Default Settings	27
DHCP Server.	19
Dial Plan	15
Dimensions	1
Direct call.	10
Display Name	21
Do not Disturb	11,13
Domain Server.	21
Don't show caller ID	14
DTMF.	21
DTMF Setting	21

E

Echo Compensation	2,4
Example: Direct IP Call	31
Example: Point-to-Point SIP Calls.	32
Extended Troubleshooting for the integrated NAT	36

F

Factory Default Setting	21,27
Factory setting	8
Factory Setting.	27
Fax setting	11
Fax Settings (T.38 (FAX) Setting	18
Features	2,4
Flash	11
Flash Key	11
Flash Time	11,17
Flash Time Setting	17

H

Handset.	13
Handset Gain	13
Handset Volume.	13
Hotline Setting.	10

I

Installation	3,5
Installing the Telephony Adapter.	1
IP Type	19

L

LAN	3,5,19
LAN Mode	18
LAN Setting	19
LAN Settings	19
LAN-port	3,5
LED Indicators and Connectors	3,5

M

Maintenance and Care	1
Making Calls through a SIP-Provider Using the TA	33
Making telephone calls	
resetting the TA	8
Making Telephone Calls	
Accepting a Waiting Cal	7
Accepting Calls	6
Broker's Call	7
Direct Call	7
Enable Call Waiting	7
Inquiry Call	7
Making External Telephone Calls	6
Room Inquiry	7
Speed Dialing from the Phone Book	6
Three-Party Conferencing	7

Mic volume.	13
My Computer	36

N

Net	18
Network	18
Network Status	18
New password	24
New username.	24
No Call Signaling by time.	13
No Ringing.	13
No Transmission of Caller ID	14
Number Replacement.	15

O

Other Settings	23
Outbound Proxy.	21

P

Package Contents	2,4
Password.	9
PC- LAN Port	3
PHONE	3,5
Phone book.	10
Phone port.	3
Phone port 1.	5
Phone port 2.	5
Port	21
Port Setting.	21
POWER	3,5
Power supply	1
Power Supply	5
Power Supply Block	1
PPPoE.	19
Prefix Unset Plan	15
Proxy Server	21

R

Reboot	27
Register Name	21
Register Password	21
Replace rule	15

S

safety notes	1
Save Changes.	25
Saving Changes	25
Set Up Call Forwarding	12
Setting for using Flash interval	17
Settings	11,23
Short Reference Manual	37
Signaling Format	14
SIP.	21
SIP Configuration.	3,5
SIP Settings.	21
SIP-Authentication	2,4
SNTP	12
SNTP Settings	11
Speed dial.	10
Speed Dial Settings	10
Status	18
STUN Setting	22

STUN settings	22
Symbol Reference.	6
Sync. Time	12
Synchronization Time	12
System Information	9

T

T.38 (FAX) Setting.	11,18
Technical specifications	1
Temperature range.	1
Time Server	12
Time Synchronization	12
Time Synchronization Setup (SNTP Settings)	12
Time Zone	12
Time Zone	12
Troubleshooting the Web-based Configuration.	36

U

Unconditional Call Forwarding	11
Update	25
User Name	9,21

User Password	24
-------------------------	----

V

V101 SIP VoIP Telephony Adapter.	2,3
V102 SIP VoIP Telephony Adapter.	4,5
VLAN	20
VLAN Settings	20
Voice codec	2,4
VoIP Protocol	2,4
Volume	11
Volume control	13
Volume Settings	11,13

W

WAN port	5
WAN settings	18
WAN Settings	18
Warm start.	27
Weight	1
Windows 2000	36

**Konformitätserklärung gemäß dem Gesetz über Funkanlagen und
Telekommunikationsendeinrichtungen (FTEG) und der Richtlinie 1999/5/EG (R&TTE)**

*Declaration of Conformity appropriate to the law of radio and telecom terminal equipment (FTEG) and
Directive 1999/5/EC (R&TTE)*

*Déclaration de conformité selon la loi sur les équipements hertziens et les équipements terminaux de télécommunication (FTEG) et la
directive 1999/5/CE (R&TTE)*

Hersteller / Verantwortliche Person:

Manufacturer / responsible person:

Funkwerk Enterprise Communications GmbH

Fabricant / personne responsable:

erklärt, dass das Produkt (Typ):

funkwerk V101

declares that the product:

funkwerk V102

déclare que le type de produit:

Telekommunikationsendeinrichtung

Telecommunications terminal equipment

Équipement de terminal de télécommunication

Verwendungszweck:

**A adapter zum Anschluss von analogen Endgeräten wie z.B.
Telefon oder Fax an ein Computer-Netzwerk (LAN)**

Intended use:

**A adapter for connecting analog terminal devices such as telephones
or fax devices to a computer network (LAN).**

Le but poursuivi:

**bei bestimmungsgemäßer Verwendung den grundlegenden Anforderungen des § 3 und den übrigen einschlägigen
Bestimmungen des FTEG (Artikel 3 der R&TTE) entspricht.**

*complies with the appropriate essential requirements of the FTEG (Article 3 of the R&TTE) and the other relevant provisions,
when used for its intended purpose.*

est conforme aux exigences fondamentales du paragraphe 3 du FTEG (article 3 du R&TTE) et des autres clauses s'y rapportant.

Gesundheit und Sicherheit gemäß § 3 (1) 1. (Artikel 3 (1) a)

Health and Safety requirements contained in § 3 (1) 1. (Article 3 (1) a)

Santé et sécurité conformes au paragraphe 3 (1) 1 (Article 3 (1) a)

harmonisierte Normen

harmonised standards

Normes harmonisées

EN60950-1: 2001

(NSR 73/23/EWG & 93/68/EWG)

(LVD 73/23/EC & 93/68/EC)

**Directive Basse Tension 73/23/CE &
93/68/CE)**

Schutzanforderungen in Bezug auf die elektromagn. Verträglichkeit § 3 (1) 2, Artikel 3 (1) b)

Protection requirements with respect to electromagn. compatibility § 3 (1) 2, (Article 3 (1) b)

Exigences des protection concernant la compatibilité électromagnétique, paragraphe 3 (1) 2, article 3 (1) b)

harmonisierte Normen

harmonised standards

Normes harmonisées

EN 55022: 1998 + A1: 2000 + A2: 2003

EN 55024: 1998 + A1: 2000 + A2: 2003

EN 61000-3-2: 2000

EN61000-3-3: 1995 + A1: 2000

(EMV-RL 89/336/EWG)

(EMCD 89/336/EC)

(Directive EMC 89/336/CE)

Anschrift

Address

Adresse

+ TK-Nr. / N° de télécommunication

Fon:

Fax:

Funkwerk Enterprise Communications GmbH

Südwestpark 94

D-90449 Nürnberg

+49 911 96 73-0

+49 911 688 07 25



Peine, 22.12.2006

Ort, Datum

Place & date of issue

Lieu et date

Dipl.-Ing. Peter Schumann, Konformitätsbeauftragter

Name und Unterschrift

Name and signature

Nom et signature

For information on support and service offerings
please visit our Website at:
www.funkwerk-ec.com
There, you will find a Service / Support area.
Phone: +49 911 9673 0

Funkwerk Enterprise Communications GmbH
Suedwestpark 94
D-90449 Nuremberg

Subjekt to modifications

Ausgabe 1

110107