

## Information on operating Gigaset VoIP telephones with Network Address Translation (NAT) routers

In general no special telephone or router configuration is required when operating a Gigaset VoIP phone with a NAT router. The configuration settings described in this section are only necessary if you encounter one of the following problems.

### Typical problems caused by NAT

- No incoming calls are possible via VoIP. Calls to your VoIP phone number are not put through.
- Outgoing calls via VoIP are not connected.
- A connection is established with the other party, but you cannot hear them and/or they cannot hear you.

### Possible solution

1. Change the port numbers of the communication ports (SIP and RTP ports) on your telephone.
2. In some cases, you must also define port forwarding for the telephone's communication ports on the router.

### 1. Changing the port numbers for SIP and RTP on your VoIP phone

On your VoIP telephone, define different (local) port numbers for the SIP and RTP ports (between 1024 and 49152).

- These numbers must not be used by any other application or host in the LAN and
- be considerably higher or lower than the SIP and RTP port numbers that you usually use (and are preset on the phone).

This procedure is particularly useful if additional VoIP phones are connected to the router.

#### To change the SIP and RTP port numbers on your VoIP phone, proceed as follows:

- Connect your PC's browser to the Web configurator of the telephone and log in.
- Open the Web page **Settings - Telephony - Advanced VoIP Settings** and change the settings for the SIP and RTP ports.

To help you remember the new port numbers (e.g. for router configuration), you can choose numbers that are very similar to the standard settings, e.g.

<b>SIP port</b>	49060	instead of	5060
<b>RTP port</b>	49004 to 49008	instead of	5004 to 5008

The screenshot shows the 'Advanced Settings' page for a Gigaset VoIP phone. The 'Listen ports for VoIP connections' section is highlighted with a red oval. In this section, the 'Use random ports' option is set to 'No'. The 'SIP port' is set to 49060, and the 'RTP port' is set to 49004 - 49008. Other settings visible include 'Transfer Call by On-Hook' (No), 'Derive target address' (from SIP contact header), and 'Hook Flash (R-key)' (disabled).

- Save the changes on your telephone.
- Wait for the active VoIP connections to be re-registered. To do so, switch to the Web page **Settings - Telephony - Connections** to see the **Status** of your VoIP connections.
- Check to see whether the problem persists. If it does, **perform step 2**.

## 2. Setting port forwarding on the router

To ensure that your specified SIP and RTP port numbers are used on the WAN interface with the public IP address, you must define port forwarding rules for the SIP and RTP ports on the router.

### To define port forwarding on the router, proceed as follows:

The terms used in the following can vary from router to router.

To forward a port, you must make the following specifications (example):

Protocol	Public port	Local port	Local host (IP)	
UDP	49060	49060	192.168.2.10	for SIP
UDP	49004–49008	49004–49008	192.168.2.10	for RTP

Local IP-Address of  
your phone in the LAN