

# One way audio problems with VoiceBlue

First, check if you use the latest firmware version (available on [www.2n.cz](http://www.2n.cz) site) and if you have an active licence by accessing the gateway via telnet and executing the commands AT13 and AT14.

In case you have older version please upgrade the firmware

## If you are using VoiceBlue with PBX

Make a direct call from your PC or SIP phone to VoiceBlue. You can use soft phone SJ phone and the called number should be in this format: [sip:number@ipadressofVoiceBlue](mailto:sip:number@ipadressofVoiceBlue). If it's working, the problem is in the configuration of your PBX.

## Check NAT configuration

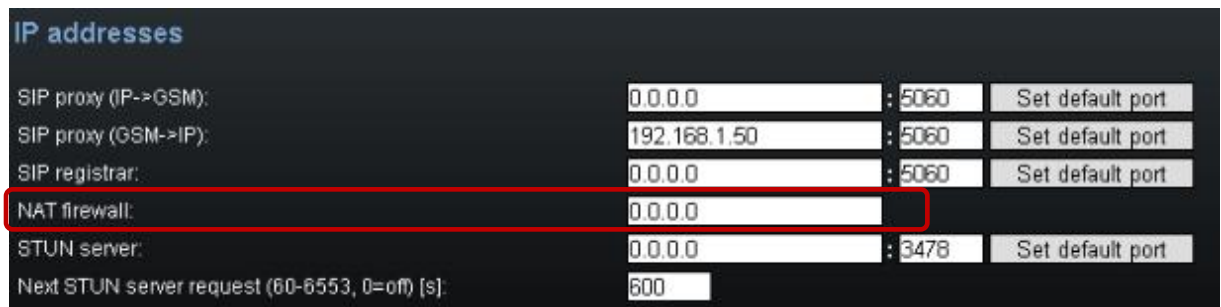
Go to Gateway configuration->VoIP Parameters

Check which RTP ports are being used by VoiceBlue (default set: 8000-8998)



Registration domain (realm):	
Caller ID:	
Username:	
Password:	
<b>Voice parameters</b>	
First RTP port (even: 1024 - 65524):	8000
Last RTP port (even: first+10 - 65534):	8998

Check the IP address of NAT firewall



<b>IP addresses</b>			
SIP proxy (IP->GSM):	0.0.0.0	: 5060	Set default port
SIP proxy (GSM->IP):	192.168.1.50	: 5060	Set default port
SIP registrar:	0.0.0.0	: 5060	Set default port
NAT firewall:	0.0.0.0		
STUN server:	0.0.0.0	: 3478	Set default port
Next STUN server request (60-6553, 0=off) [s]:	600		



Access to **UDP port** translation page of the configurator of your router and look for 5060 and RTP ports. This ports have to be opened.

Single Port Forwarding					
Application Name	External Port	Internal Port	Protocol	To IP Address	Enabled
sip	5060	5060	Both	192.168.50.23	<input checked="" type="checkbox"/>

Port Range Forwarding					
Application Name	Start ~ End Port	Protocol	To IP Address	Enabled	
RTP	8000 to 8988	UDP	192.168.50.24	<input checked="" type="checkbox"/>	

Example of router *port forwarding* configuration

If you have a public IP check if RTP packets are not being sent out of your network by NAT translation

## Check LAN configuration

Make a wireshark trace on your PC during the *one way audio* call from softphone. Once done, check if the Ethernet information contains **2nTelekomu** as source or destination.

Then, check the if Connection information (i.e IP address for RTP ports) is correct for your scenario. If not, revise settings of your router.

```
4 0.549517 192.168.22.42 192.168.22.37 SIP/SDP Request: INVITE sip:3772@192.168.22.37:5063 SIP/2.0
5 0.565587 192.168.22.37 192.168.22.42 SIP Status: 100 Trying
6 0.582489 192.168.22.37 192.168.22.42 SIP Status: 180 Ringing
7 4.681847 192.168.22.37 192.168.22.42 SIP/SDP Status: 200 OK, with
8 4.686114 192.168.22.42 192.168.22.37 SIP Request: ACK sip:3772@192.168.22.37:5063 SIP/2.0
9 4.697455 192.168.22.37 192.168.22.42 RTP PT=ITU-T G.711 PCMU, 8000 Hz, 10ms, 1024 samples
10 4.878966 192.168.22.37 192.168.22.42 RTP PT=ITU-T G.711 PCMU, 8000 Hz, 10ms, 1024 samples
11 4.880458 192.168.22.37 192.168.22.42 RTP PT=ITU-T G.711 PCMU, 8000 Hz, 10ms, 1024 samples
12 4.880491 192.168.22.37 192.168.22.42 RTP PT=ITU-T G.711 PCMU, 8000 Hz, 10ms, 1024 samples
13 4.881925 192.168.22.37 192.168.22.42 RTP PT=ITU-T G.711 PCMU, 8000 Hz, 10ms, 1024 samples
14 4.883341 192.168.22.37 192.168.22.42 RTP PT=ITU-T G.711 PCMU, 8000 Hz, 10ms, 1024 samples
15 4.883926 192.168.22.37 192.168.22.42 RTP PT=ITU-T G.711 PCMU, 8000 Hz, 10ms, 1024 samples
16 4.887826 192.168.22.37 192.168.22.42 RTP PT=ITU-T G.711 PCMU, 8000 Hz, 10ms, 1024 samples
```

Frame 4 (744 bytes on wire, 744 bytes captured)

- Ethernet II, Src: 2nTelekomu\_00:b9 (00:50:c2:9b:f0:b9), Dst: Vmware\_a4:6e:96 (00:0c:29:a4:6e:96)
  - Destination: Vmware\_a4:6e:96 (00:0c:29:a4:6e:96)
  - Source: 2nTelekomu\_00:b9 (00:50:c2:9b:f0:b9)
- Type: IP (0x0800)
- Internet Protocol, Src: 192.168.22.42 (192.168.22.42), Dst: 192.168.22.37 (192.168.22.37)
- User Datagram Protocol, Src Port: 5063 (5063), Dst Port: 5063 (5063)
- Session Initiation Protocol
  - Request-Line: INVITE sip:3772@192.168.22.37:5063 SIP/2.0
  - Message Header
  - Message Body
    - Session Description Protocol
      - Session Description Protocol Version (v): 0
      - Owner/Creator, Session Id (o): voiceBlueNext 39463 21584 IN IP4 192.168.22.42
      - Session Name (s): GSM call
      - Connection Information (c): IN IP4 192.168.22.42
      - Time Description, active time (t): 0 0
      - Media Description, name and address (m): audio 8002 RTP/AVP 8 0 101



## One way audio on hold

Make If you cannot hear any audio after returning to a previously held call, make sure that re invite packets contain SDP information in wireshark trace.

In case you followed all points of the document and the problem persist, write us an detailed explanation of the situation in which one way audio appears, send us configuration file of your VoiceBlue, one wireshark trace and description of how is your connection scenario including IP addresses (sketch or picture recommended)

## Links

### **Latest firmware version of VoiceBlueNext**

<http://www.2n.cz/en/products/gsm-gateways/voip/voiceblue-next/downloads/> (subtab firmware)

### **Latest firmware version of VoiceBlueLite**

<http://www.2n.cz/en/products/gsm-gateways/voip/voiceblue-lite/downloads/> (subtab firmware)

### **Guide for One way audio in Asterisk**

[http://www.asteriskguru.com/tutorials/sip\\_nat\\_oneway\\_or\\_no\\_audio\\_asterisk.html](http://www.asteriskguru.com/tutorials/sip_nat_oneway_or_no_audio_asterisk.html)