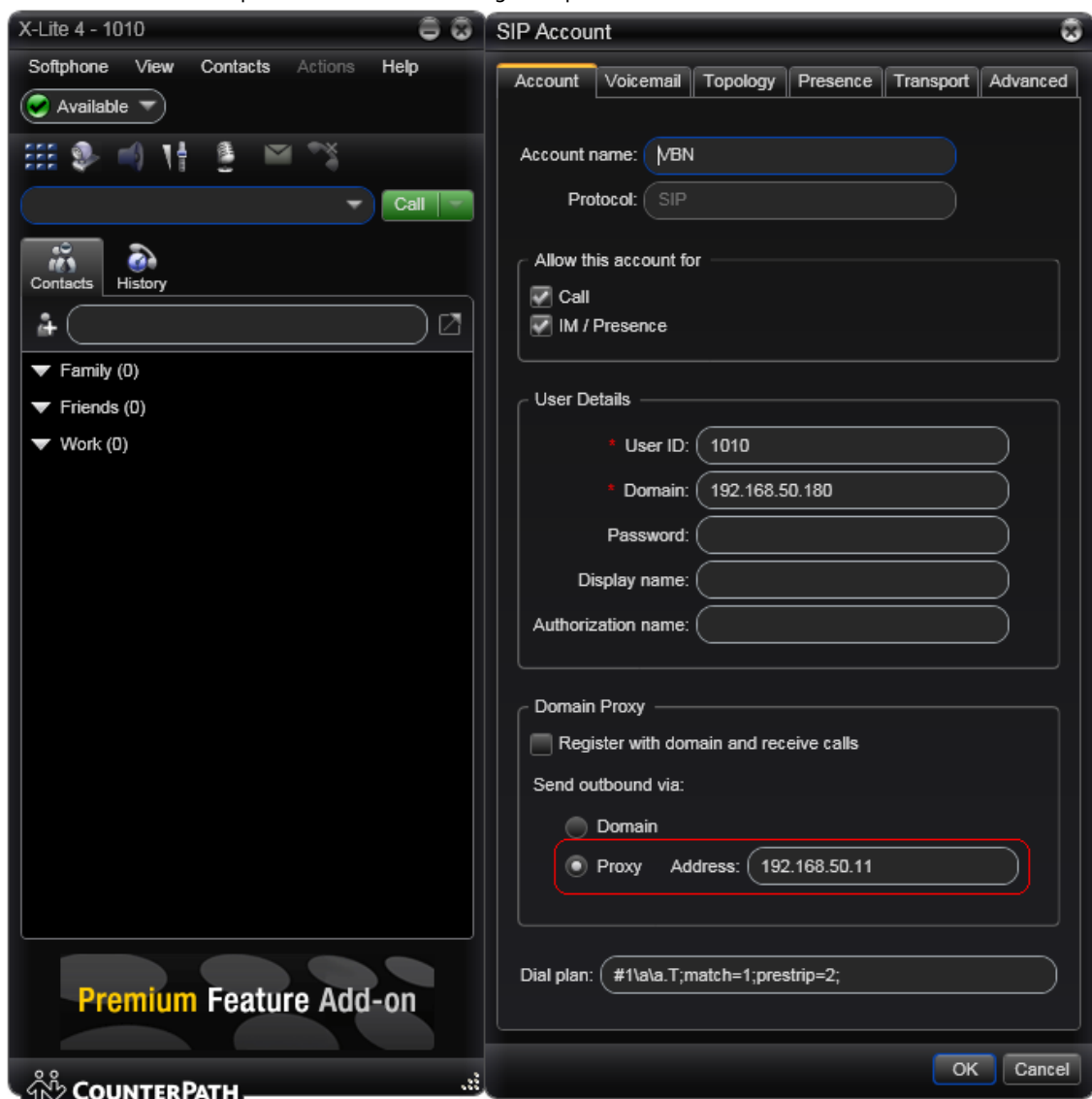


Test call - How to make a test call from Softphone to 2N® VoiceBlue MAX

- Once you have set up 2N® VoiceBlue MAX, the simplest way to test basic functionality is to make some test calls from you PC.
- In this guide you will find the instructions for making a direct call to the 2N® VoiceBlue MAX from X-Lite and Sjphone

X-Lite Configuration

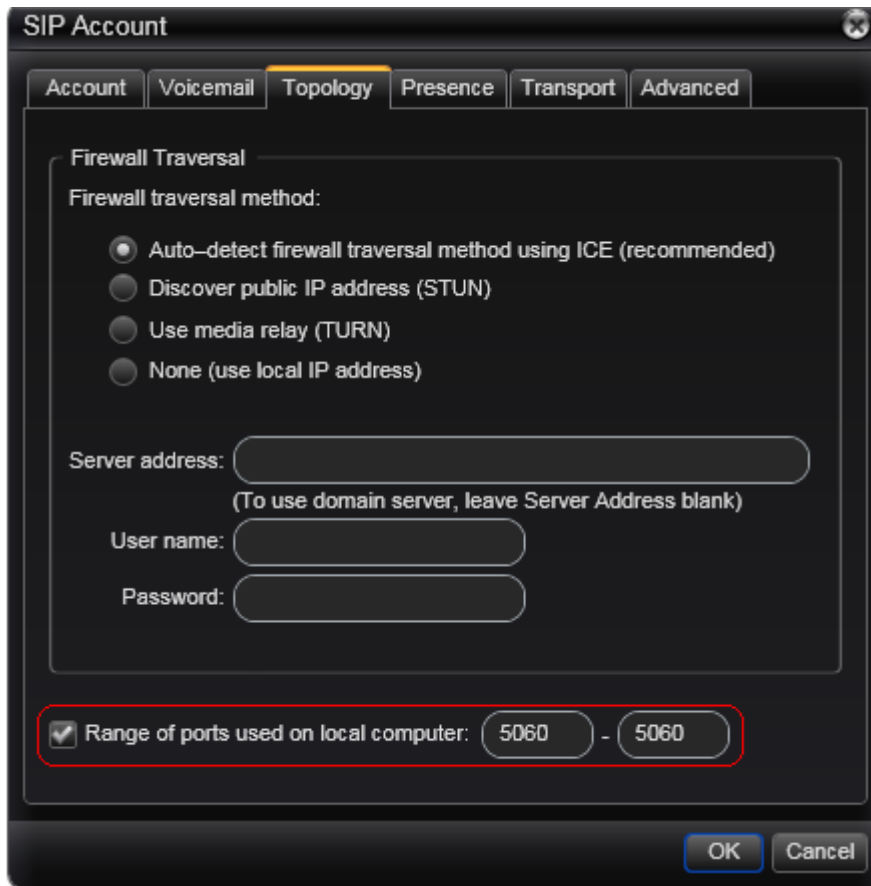
- In the X-Lite menu Softphone -> Account Setting set up the user account



- The only important thing which should be set up correctly is Proxy Address; it should be the IP address of the 2N® VoiceBlue MAX. User ID and Domain are mandatory parameters but it does not matter what you fill in these fields.
- Then just dial number without IP address.



- For incoming calls it is necessary to set up range of ports allowed for incoming SIP messages.



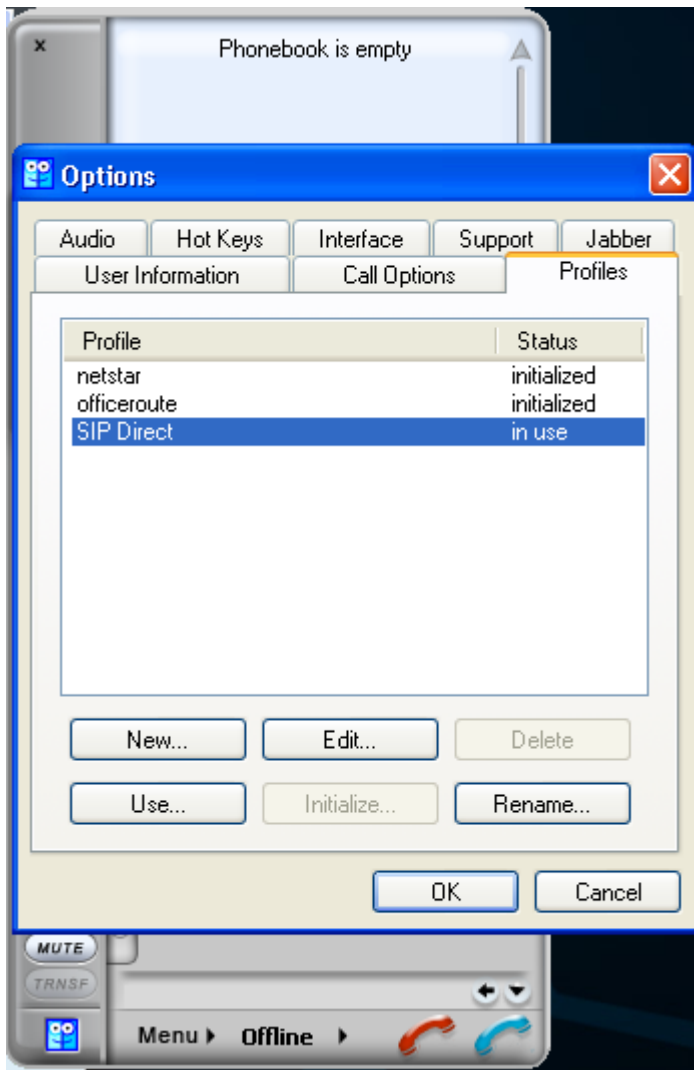
The screenshot shows the 'SIP Account' configuration window with the 'Topology' tab selected. The 'Firewall Traversal' section is active, showing the following options and fields:

- Firewall traversal method:
 - Auto-detect firewall traversal method using ICE (recommended)
 - Discover public IP address (STUN)
 - Use media relay (TURN)
 - None (use local IP address)
- Server address:
(To use domain server, leave Server Address blank)
- User name:
- Password:
- Range of ports used on local computer: -

Buttons for 'OK' and 'Cancel' are located at the bottom right of the window.

Sjphone Configuration

- In Sjphone Options -> Profiles section choose SIP Direct. This account is basically available in default configuration.

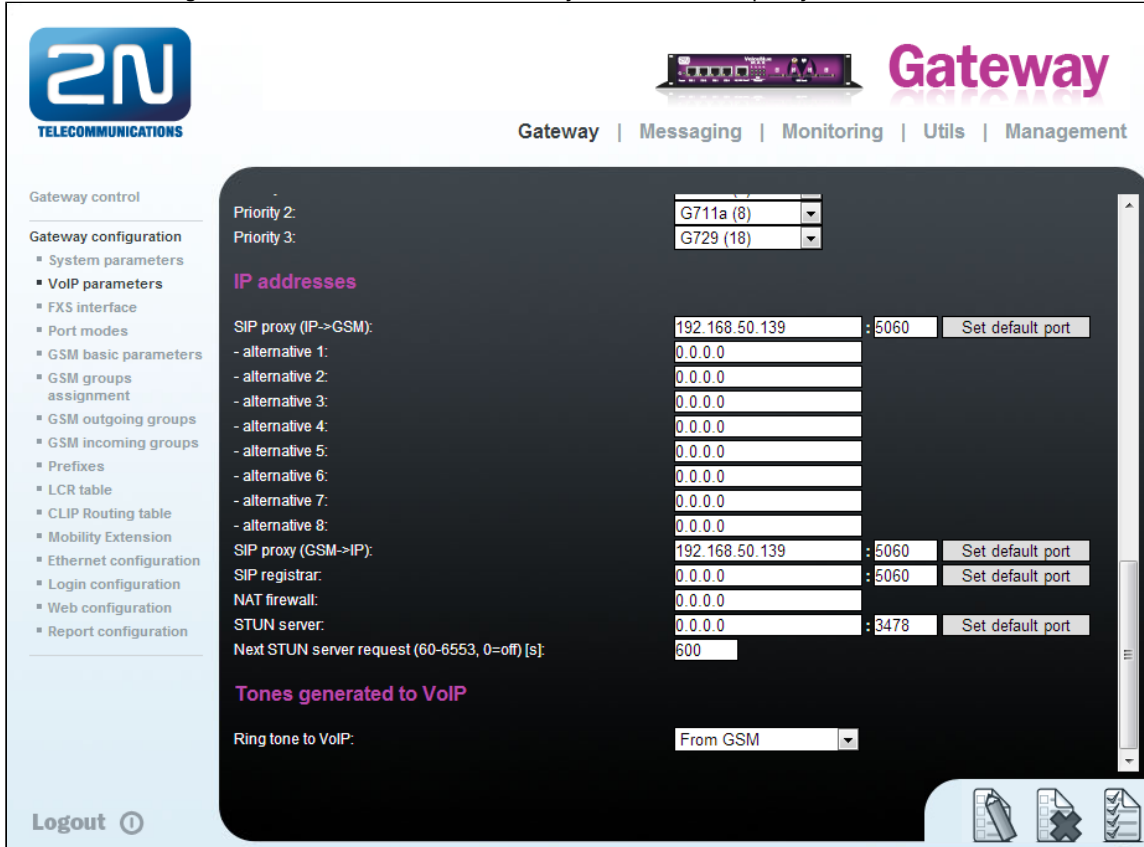


- Dial the number in format sip:called_number@VBM_ip_address



Settings on the 2N[®] VoiceBlue MAX side.

- The IP address of you PC should be inserted into the SIP Proxy list (Gateway configuration -> VoIP Parameters -> SIP proxy (IP -> GSM) -> Alternative. Note that you can fill in up to 9 IP addresses from which you can receive SIP traffic. If SIP proxy (IP -> GSM) is 0.0.0.0, 2N[®] VoiceBlue MAX can accept SIP traffic from any host.
- To allow incoming calls from GSM to IP it is necessary to fill in the SIP proxy (GSM -> IP).



The screenshot shows the 'Gateway' configuration page for a 2N device. The interface includes a navigation menu on the left with options like 'Gateway control', 'Gateway configuration', and 'Logout'. The main content area is titled 'Gateway' and contains several configuration sections:

- Priority 2:** G711a (8)
- Priority 3:** G729 (18)
- IP addresses:**
 - SIP proxy (IP->GSM):** 192.168.50.139 : 5060 [Set default port]
 - alternative 1: 0.0.0.0
 - alternative 2: 0.0.0.0
 - alternative 3: 0.0.0.0
 - alternative 4: 0.0.0.0
 - alternative 5: 0.0.0.0
 - alternative 6: 0.0.0.0
 - alternative 7: 0.0.0.0
 - alternative 8: 0.0.0.0
 - SIP proxy (GSM->IP):** 192.168.50.139 : 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
- Tones generated to VoIP:** Ring tone to VoIP: From GSM

More product information:

2N[®] VoiceBlue MAX (Official Website 2N)