Works with incoming and outgoing SIP calls.

Note that personal-voip.de wants your SIP ID as username, not your SIP number!

Also note that the contact must be send as <sipID>@sip.personal-voip.de. Therefore the parameter extension-in-contact must be set to true and the extension param must be empty or set to the same value as username!

```
<include>
    <gateway name="personal-voip">
        <param name="username" value="ONE OF YOUR SIP IDs"/>
        <param name="password" value="your_password"/>
        <param name="extension" value="comment this param out or set it to the same sip id you out in the username field. Do not put anything else here!"/>
        <param name="ping" value="if you are behind a route and experience problems set this to a smaller value. Otherwise comment it out"/>
        <param name="proxy" value="sip.personal-voip.de"/>
        <param name="register" value="true"/>
        <param name="extension-in-contact" value="true"/>
        <param name="caller-id-in-from" value="false"/>
    </gateway>
</include>
```

If you discover that no/not the right caller ID is displayed on outgoing calls this might be helpful:

Fore some reason personal-voip.de seems to ignore effective_caller_id_number and origination_caller_id_number but instead uses the value from effective_caller_id_name and origination_caller_id_name. If the value from the name field is numeric only the correct number is displayed. Otherwise the the first number from your block of phonenumbers is used.