

FreePBX (Trixbox)
configuration
for GoAndCall.com VoIP services.

(GoAndCall.com VoIP services are provided by
XeloQ Communications - for more information look at
<http://www.xeloq.eu>)

Go to Tools -> Module admin and enable modules you need. 'Core' module is required, everything else is optional.

Sample Screenshot:



Module Admin

Module Administration

[Connect to Online Module Repository](#)

Enabled Modules

| | Module | Version | Type | Category |
|--------------------------|-------------|---------|-------|----------|
| <input type="checkbox"/> | Core (core) | 1.1 | setup | Basic |

Disable Selected

Disabled Modules

| | Module | Version | Type | Category |
|--------------------------|---|---------|-------|-----------------|
| <input type="checkbox"/> | Asterisk API (manager) | 1.0.2 | tool | Basic |
| <input type="checkbox"/> | Feature Code Admin (featurecodeadmin) | 1.0 | setup | Basic |
| <input type="checkbox"/> | Follow Me (findmefollow) | 1.2.5 | setup | Basic |
| <input type="checkbox"/> | Time Conditions (timeconditions) | 2.1 | setup | Basic |
| <input type="checkbox"/> | Call Forward (callforward) | 1.0.2 | setup | Call Management |
| <input type="checkbox"/> | Call Waiting (callwaiting) | 1.0.1 | setup | Call Management |
| <input type="checkbox"/> | SellVOIP (sellvoip) | 0.1 | setup | ITSP |
| <input type="checkbox"/> | VoicePulse (voicepulse) | 0.1 | setup | ITSP |
| <input type="checkbox"/> | Voicemail (voicemail) | 1.0 | setup | Messaging |
| <input type="checkbox"/> | Conferences (conferences) | 1.0.1 | setup | Module |
| <input type="checkbox"/> | IVR (ivr) | 2.2.6 | setup | Module |

Enable Selected

Not Installed Local Modules

| | Module | Version | Type | Category |
|--------------------------|--------------------------|---------|-------|----------|
| <input type="checkbox"/> | Rina Groups (rinaaroups) | 1.2.2 | setup | Basic |

Then you need to add SIP trunk at the *Setup -> Trunks -> Add SIP Trunk*.

Outbound Caller ID: 7XXXXX

Trunk Name: goandcall (or anything else what you like)

Peer details:

```
allow=g729&g723.1
canreinvite=no
disallow=all
host=sip.goandcall.com
type=peer
username=7XXXXX
secret=YourPassword
```

If you plan to use inbound calling, you need to add **Register String:**

7XXXXX:YourPassword@sip.goandcall.com

Sample Screenshot:

The screenshot shows the FreePBX web interface for adding a SIP trunk. The navigation menu includes Setup, Tools, Reports, Panel, and Recordings. The left sidebar lists various system components, with 'Trunks' selected. The main content area is titled 'Add SIP Trunk' and contains the following sections:

- General Settings:** Outbound Caller ID: 7XXXXX, Maximum channels: [empty].
- Outgoing Dial Rules:** Dial Rules: [empty], Clean & Remove duplicates button, Dial rules wizards: (pick one) [dropdown], Outbound Dial Prefix: [empty].
- Outgoing Settings:** Trunk Name: [empty], PEER Details: allow=g729&g723.1, canreinvite=no, disallow=all, host=sip.goandcall.com, type=peer, username=7XXXXX, secret=YourPassword.
- Incoming Settings:** USER Context: [empty], USER Details: [empty].
- Registration:** Register String: 7XXXXX:YourPassword@sip.goandcall.com, Submit Changes button.

After this you need to add Outbound route under *Setup -> Outbound Routes*:

Route Name: pstn (or anything else)

Dialing Patterns:
00.

Trunk Sequence:
SIP/goandcall

treePBX

•Setup •Tools •Reports •Panel •Recordings

Setup

| | | |
|------------------------|--|------------------|
| Administrators | <h3>Add Route</h3> <p>Route Name: <input type="text" value="pstn"/></p> <p>Route Password: <input type="text"/></p> <p>Emergency Dialing: <input type="checkbox"/></p> <p>Dial Patterns</p> <div style="border: 1px solid #ccc; padding: 5px; min-height: 40px;">00.</div> <p style="text-align: center;"><input type="button" value="Clean & Remove duplicates"/></p> <p>Insert: <input type="button" value="Pick pre-defined patterns"/></p> <p>Trunk Sequence</p> <p><input type="text" value="SIP/goandcall"/> <input type="button" value="Add"/></p> <hr/> <p style="text-align: center;"><input type="button" value="Submit Changes"/></p> | Add Route |
| Extensions | | 0 pstn |
| General Settings | | |
| Inbound Routes | | |
| Outbound Routes | | |
| Trunks | | |

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You also need to configure each your extension to allow G.729 or G.723 codec by adding:

Allow: g729&g723.1

to each extension configuration under Setup -> Extension -> Name of your Extension (right side):



•Setup •Tools •Reports •Panel •Recordings

Setup

| | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
|-------------------|--|---|---|----------|--------------------------------------|-------------|---------------------------------|---------|--|------|--------------------------------------|------|-------------------------------------|-----|------------------------------------|------|-----------------------------------|---------|---------------------------------|-----------|----------------------|-------------|----------------------|----------|----------------------|-------|--|------|---------------------------------------|-------------|----------------------|---------|--|--|
| Administrators | SIP Extension: 2000 Delete Extension 2000 Edit Extension <hr/> Display Name: <input type="text" value="Phone 1"/> Extension Options <hr/> Direct DID: <input type="text"/> DID Alert Info: <input type="text"/> Outbound CID: <input type="text"/> Emergency CID: <input type="text"/> Record Incoming: <input type="text" value="On Demand"/> Record Outgoing: <input type="text" value="On Demand"/> Device Options <hr/> <table><tr><td>secret</td><td><input type="text" value="LocalPhonePassword"/></td></tr><tr><td>dtmfmode</td><td><input type="text" value="rfc2833"/></td></tr><tr><td>canreinvite</td><td><input type="text" value="no"/></td></tr><tr><td>context</td><td><input type="text" value="from-internal"/></td></tr><tr><td>host</td><td><input type="text" value="dynamic"/></td></tr><tr><td>type</td><td><input type="text" value="friend"/></td></tr><tr><td>nat</td><td><input type="text" value="never"/></td></tr><tr><td>port</td><td><input type="text" value="5060"/></td></tr><tr><td>qualify</td><td><input type="text" value="no"/></td></tr><tr><td>callgroup</td><td><input type="text"/></td></tr><tr><td>pickupgroup</td><td><input type="text"/></td></tr><tr><td>disallow</td><td><input type="text"/></td></tr><tr><td>allow</td><td><input type="text" value="g729&g723.1"/></td></tr><tr><td>dial</td><td><input type="text" value="SIP/2000"/></td></tr><tr><td>accountcode</td><td><input type="text"/></td></tr><tr><td>mailbox</td><td><input type="text" value="2000@device"/></td></tr></table> Voicemail & Directory: <input type="text" value="Disabled"/> | secret | <input type="text" value="LocalPhonePassword"/> | dtmfmode | <input type="text" value="rfc2833"/> | canreinvite | <input type="text" value="no"/> | context | <input type="text" value="from-internal"/> | host | <input type="text" value="dynamic"/> | type | <input type="text" value="friend"/> | nat | <input type="text" value="never"/> | port | <input type="text" value="5060"/> | qualify | <input type="text" value="no"/> | callgroup | <input type="text"/> | pickupgroup | <input type="text"/> | disallow | <input type="text"/> | allow | <input type="text" value="g729&g723.1"/> | dial | <input type="text" value="SIP/2000"/> | accountcode | <input type="text"/> | mailbox | <input type="text" value="2000@device"/> | Add Extension Phone 1 <2000> |
| secret | | <input type="text" value="LocalPhonePassword"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| dtmfmode | | <input type="text" value="rfc2833"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| canreinvite | | <input type="text" value="no"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| context | | <input type="text" value="from-internal"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| host | | <input type="text" value="dynamic"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| type | <input type="text" value="friend"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| nat | <input type="text" value="never"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| port | <input type="text" value="5060"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| qualify | <input type="text" value="no"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| callgroup | <input type="text"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| pickupgroup | <input type="text"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| disallow | <input type="text"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| allow | <input type="text" value="g729&g723.1"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| dial | <input type="text" value="SIP/2000"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| accountcode | <input type="text"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| mailbox | <input type="text" value="2000@device"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Extensions | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| General Settings | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Inbound Routes | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Outbound Routes | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Trunks | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |

