

Interop Hybird 120 Gigaset Edition - Sipcall Business

Sipcall Business

Feature	
Outgoing Calls	Yes
Incoming Calls	Yes
CLIP incoming	Yes
CLIP outgoing	Yes
Call Forwarding	Yes
Call Transfer	Yes
Call Waiting	Yes
DTMF	Yes
Anonymous Call	Yes*
A-number forwarding	Yes

* Must be activated by Sipcall. Only for permanent anonymous calling available.

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Hybird 120 Gigaset Edition settings.

In the Hybird 120 Gigaset Edition go to "Assistants" - "PBX" and click on "New": --> add a new SIP Provider

hybird 120 Gigaset

Hybird_120_GE

LanguageEnglish

ViewStandard

Online Help

Save configuration

TrunksUsers

Assistants

First steps

Internet Access

VPN

PBX

System Management

Physical Interfaces

VoIP

Numbering

Terminals

Call Routing

Applications

LAN

Networking

Multicast

WAN

VPN

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Local Services

Maintenance

External Reporting

Monitoring

View20per pageGo

No.	Name	Connection Type	Ports	Status		
01	Sipcall	SIP Provider (DDI)	Sipcall			



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New



















Provider settings:

Give the Line an unique name, insert the Registration user name and Password.


hybird 120 Gigaset
Hybird_120_GE

Language English  View Standard  Online Help

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Trunks **Users**

 **Warning: Country code and/or City code not configured!**

SIP Provider Settings

Name

Access Type

Direct Dial-In

Authentication ID

Password

User Name

Registrar


Trunk Numbers


Base Number

Class of Service

Class of Service

CoS Default





Add

Advanced Settings

OK

Cancel

Just add the telephone numbers for this SIP trunk in the "Advanced Settings"

Advanced Settings

Registrar

Registrar Port

5060

Transport Protocol

☒ UDP
☐ TCP

STUN server

STUN server

Port STUN server

3478

Trunk Numbers

P-P DDI Exception

P-P DDI Exception	Displayed Name	
5680	0325	
5681	0325	
5682	0325	
5683	0325	
5684	0325	

Add

Further Settings

Generate international phone number

☐ Enabled

Generate national subscriber number

☐ Enabled

OK

Cancel

Re

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VoIP settings:

Here you can define how the outgoing lines have to look like (e.g. Clip no Screening)

Save configuration

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SIP Provider

Locations

Codec Profiles

Options

Basic Settings

Description

Sipcall

Provider Status

☒ Active
 ☐ Inactive

Access Type

☐ Single Number(s)
 ☒ Direct Dial-In

Authentication ID

41325135680

Password

••••••••

User Name

41325135680

Domain

Outgoing Signalisation Settings

Outgoing Signalisation

Individual CLIP no Screening Number

Signal remote caller number

☒ Enabled

Registrar

Registrar

pro1.voipgateway.org

Registrar Port

5060

Transport Protocol

☒ UDP
 ☐ TCP

STUN

STUN server

Port STUN server

3478

Timer

Registration Timer

60 Seconds

Advanced Settings

VoIP Menu Advanced settings:

Please choose here the below settings when you use the CLIP No Screening.

Advanced Settings

Proxy	<input type="text"/>
Proxy Port	<input type="text" value="5060"/>
Transport Protocol	<input checked="" type="radio"/> UDP <input type="radio"/> TCP
Further Settings	
From Domain	<input type="text"/>
Number of allowed simultaneous Calls	<input type="text" value="No Limitation"/>
Location	<input type="text" value="Any Location"/>
Codec Profiles	<input type="text" value="System Default"/>
Dial End Monitoring Time	<input type="text" value="5"/> Seconds
Call Hold inside the PBX system	<input checked="" type="checkbox"/> Enabled
Call Forwarding extern (SIP 302)	<input type="checkbox"/> Enabled
Generate international phone number	<input type="checkbox"/> Enabled
Generate national subscriber number	<input type="checkbox"/> Enabled
Deactivate number suppression	<input type="checkbox"/> Enabled
SIP Header Field for User Name	<input type="radio"/> P-Preferred <input type="radio"/> P-Asserted <input checked="" type="radio"/> None
SIP Header Field(s) for Caller Address	<input type="checkbox"/> Display
	<input checked="" type="checkbox"/> User Name
	<input type="checkbox"/> P-Preferred
	<input type="checkbox"/> P-Asserted
Substitution of International Prefix with "+"	<input type="checkbox"/> Enabled
PBX coupling	<input type="checkbox"/> Enabled
Delete SIP bindings after Restart	<input checked="" type="checkbox"/> Enabled
Upstreaming Device with NAT	<input type="checkbox"/> Enabled
Early media support	<input checked="" type="checkbox"/> Enabled
Provider without Registration	<input type="checkbox"/> Enabled