DIVUS-VS

DIVUS-VS User Manual

Version 2.0

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GENERAL INFORMATION

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Please read the manual before beginning and keep the manual for later use.

The manual has been conceived and written for users who are experienced in the use of PCs and automation technology.

CONVENTIONS

[KEYS]	Keys that are to be pressed by the user are given in square brackets, e.g. [CTRL] or [DEL]
COURIER	On-screen messages are given in the Courier font, e.g. C:\>
COURIER BOLD	Keyboard input to be made by the user are given in Courier bold, e.g. C:\>**DIR**
„…"	Names of buttons to be pressed, menus or other onscreen elements and product names are given within double quotes. (e.g. "Configuration").
PICTOGRAMS	In this manual the following symbolic are used to indicate particular text blocs.
⚠	*Caution!* A dangerous situation may arise that may cause damage to material.
ⓘ	*Hint* Hints and additional notes
NEW	*New* New features

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1   Introduction

1.1   PREFACE

Divus offers a family of devices which can be adopted as VoIP servers for small to very large intercom and door opening systems, called DIVUS-VS.

With this products you can build systems with up to 560 participants. This manual will explain how to configure a DIVUS-VS successfully.

The following table shows the different models in comparison:

MODEL	INTERNAL UNITS	EXTERNAL UNITS	CONCURRENT CALLS
DIVUS-VS1	20	10	15
DIVUS-VS2	60	10	35
DIVUS-VS3		560[1]	280

DIVUS-VS uses the wide spread SIP protocol for VoIP communication. SIP uses the UDP network protocol for the communication between server and clients.

The SIP standard allows the usage of a wide range of codecs to pack and transport the voice data. To have a good speech quality it is recommended to use these codecs: *ulaw/PCMU* and/or *alaw/PCMA*. These codecs need a bandwidth of 64kBit/s but offer the best possible quality.

Participants of such a system may be called SIP clients. These SIP clients can connect to the DIVUS-VS server over the network using so called SIP accounts; these accounts are configured on the DIVUS-VS and the data entered there must then also be used to configure each of the SIP client devices.

Each call number, calling a single SIP client or a group of clients (group call) is also configured on the DIVUS-VS. If a SIP client calls an existing number, the call is processed by the DIVUS-VS and forwarded to a single or multiple recipients according to the configuration.

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[1] This is the maximum sum of indoor and outdoor units, which DIVUS-VS3 can handle. DIVUS-VS3 has no standardized pre-configuration, the indoor units, outdoor units, direct and group calls are configured project-specifically before delivery.

1.2   DEFAULT CONFIGURATION

To make its use as simple as possible, the DIVUS-VS comes with a set of SIP accounts and call numbers which can be immediately used without any configuration intervention.

It's important to distinguish between *SIP accounts* and *call numbers*.

A **SIP account** is a way for a single SIP client to register at the DIVUS-VS: only registered devices can be called and take part in the VoIP system.

A **call number** represents a call destination which is reached when a SIP client dials that number. A call number may stand for a single SIP client or a group of SIP clients called at the same time. The term *extension* also defines a call number.

The default configuration has a call number for each SIP account, which is identical to the identifier of the SIP account. So when using the default configuration, each registered SIP client can be reached at the number which represents him and which is shown in the DIVUS-VS status page.

As DIVUS-VS is primarily built and optimized for door communication and for intercom, there is a distinction in the SIP accounts between *internal units* and *external units*. Internal units are meant to be used for all the devices inside the building which get calls from the outdoor devices. External units are all the outdoor devices (door opening devices) which are installed on the outside of a building/apartment and are used to ring up the devices inside.

SIP ACCOUNTS	PURPOSE
101, 102, 103 ...	The default configuration already has a set of SIP accounts for internal units which may be directly used without any changes. The numbering of those accounts starts from 101.
901, 902, 903 ...	The default configuration already has a set of SIP accounts for external units which may be directly used without any changes. The numbering for external units starts from 901.

To ease the setup of single SIP clients, SIP accounts in the default configuration use the same 3 digit number for ID and password as for the numbering of the accounts themselves. Therefore e.g. the SIP account **101** would use ID „**101**" and password „**101**" to configure its SIP client.

Call numbers are divided into those for internal devices, those for external devices and those for group calls:

CALL NUMBER	PURPOSE
101,102, 103 ... 901, 902, 903 ...	The call number here always corresponds to the SIP account. Thus calling the number, the SIP client which registered at the DIVUS-VS with that account will be called. So every SIP client can be reached directly.
dooropener0101, dooropener0201 ...	This is a special set of alphanumeric call numbers which are only meant to be used with a DIVUS Videophone-Box (which is used to transform an analogue door-opening device into a SIP capable one). These numbers are group calls calling 20 numbers each (101, 102, ..., 120)
20060101, 20060201 ...	These call numbers are pre-defined group call numbers to 20 numbers (101, 102, ..., 120) of internal devices. They are meant to be used according to the following scheme: external unit 901 button 1 calls 20060101, external unit 902 button 1 calls 20060201 etc. If the single external units have more than one button, you should implement additional group call numbers in this way: external unit 901 button 2 calls 20060102, external unit 901 button 3 calls 20060103, external unit 902 button 2 calls 20060202 etc. The group calls may be easily copied, pasted and edited for such cases. See Chapter 3.4.3 for further details.

Note: DIVUS-VS3 has no default configuration, because it is used in bigger installations, which always have special requirements for the configuration of DIVUS-VS3. Therefore the configuration of DIVUS-VS3 will be made always project-specific before delivery. Please get in touch with DIVUS for the correct configuration of the device for your project.

2   Commissioning

2.1   CABLING AND FIRST CONNECTION

Plug the DIVUS-VS onto the network and the power outlet through its cables. When the devices has booted, it can be reached at the address **https://192.168.0.100** using a web browser, getting to its web interface.

The web GUI is only reachable over an SSL-encrypted connection. Please accept the security warning or add a security exception (depending on the used web browser) to allow access to the web pages.

⚠️ **Warning**: The air circulation on the top and bottom side of the device must be guaranteed! Corresponding spacers must never be removed!

⚠️ **Warning**: DIVUS-VS contains a number of functions and parameters which can only be used or apply if you use it as network router. The DIVUS-VS is optimized for being used in the local network and should not be used as a router, but instead should be placed behind the internet router – like most other network devices.

If no network setting changes of the DIVUS-VS are necessary, the device is ready to run. If changes should be necessary, please see the next chapter of this manual.
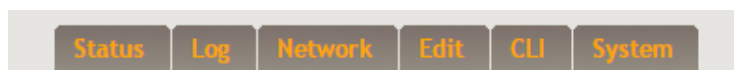
⚠️ **Warning**: It is highly recommended to change the password for the user „admin" once the commissioning is completed. See chapter 1.4.6 for further details.

3   Configuration

3.1   THE WEB INTERFACE

The web interface has a simple structure with tabs in the upper part which can be used to reach the single configuration pages.

| Status | Log | Network | Edit | CLI | System |

Through a click on the respective tab you will reach the configuration page of the web interface.

With the exception of the „Status" page, access to the pages is possible only signing in with user name and password. The default values are **admin** as user name and **1234** as password.

⚠ **Warning**: Don't forget to change the default password once you are logged in. Go to the „System" configuration page to do this.

3.1.1   STATUS

This is the page which appears first when you reach the DIVUS-VS web address. It contains general information about the current status of the device and of the VoIP system. At the first connection, for normal maintenance or for troubleshooting, here you will get a first general overview about the network settings, the memory usage, date/time settings, used SIP channels and active or predefined SIP participants. To change any of those settings, you will then change to one of the other tabs.

Further details about the meaning of the information found on the Status page can be obtained in chapter 3.3.

3.1.2   LOG

Here you'll find the settings for the advanced logging of the DIVUS-VS VoIP service activities:

- „Enable full logging": enables/disables the advanced logging

- „Verbose": sets the amount/type of logged data. The higher the value, the more is logged. The range goes from 0 (no logging) to 99 (logging all available information). Enabling and disabling the logging automatically switches between the values 99 and 4.

- „Download Log": This button will start the download of the current log file.

- „Time/Date": These may be set manually, which is useful for setups which have no internet access and can therefore not be synchronized over an external time server. On the other hand, having correct time settings is important for troubleshooting (e.g. when comparing data from different devices). You may also set the correct time zone on this page. The button „Set time/date and reboot" can then be used to store the changes and have them applied after the reboot.

The button „Save" will save the configuration changes for logging and the VoIP service will be reloaded automatically.

⚠ **Warning:** Logging should be enabled only while diagnosing a VoIP related issue. Once done, the logging function should be disabled for the normal system use.

3.1.3   NETWORK

Here you can find all the settings related to the network (IP address, gateway, DNS, Hostname etc.)

For the normal use in the 192.168.0.x network the system is already set up correctly and ready to run without any changes.

Please refer to chapter 3.2 for details on network settings.

3.1.4   EDIT

In this tab you can open all of the DIVUS-VS main configuration files in an embedded editor and apply changes. You then must restart or reload the respective service to store the changes and make them effective.

See chapter 3.4 for further details.

3.1.5   CLI

The CLI (*command line interface*) is a system terminal embedded inside a web page, offering a powerful interface to the operating system. Access to the CLI is granted only to DIVUS support personnel – the end user will not have access to it.

3.1.6   SYSTEM

On this last page you can find settings for system maintenance.

The most important functions are:

- Changing the *admin* password: insert the new password in the two fields „New Password" and „Confirm Password", then press the button „Set Password" to apply the changes.

- Backup/restore settings:

    o   Backup: Under „Configuration/File Backup" choose „Basic Configuration files" and push the button „Download Backup" to locally download the current settings (in a file).

    o   Restore: Under „Restore Basic Configuration" push the button „*Choose File*" and browse your computer to select the previously saved backup file; then pushing the „Restore Configuration" button will read the backup settings and restore the device to that state.

- System reboot: choose „now" or another desired time and set the „Confirm" checkbox. Then push the „Reboot" button which will cause the device to restart.

Warning: Restoring a device with a previously created backup will also set the network address from that backup. Therefore after the restoring the device might change network settings and be reachable only over the changed IP address!

3.2   NETWORK CONFIGURATION

⚠️ **Warning:** We recommend to leave all the additional settings not explicitly described here **unchanged** to grant a flawless functionality.

The main settings here are the following:

DOMAIN	Insert the IP address of the DIVUS-VS. Default `192.168.0.100`
DNS	Here you can insert a DNS server. Useful e.g. for the name resolution of the time server.
STATIC IPV4	Insert the IP address of the DIVUS-VS. Default `192.168.0.100`
NETMASK	Insert the netmask here. Default 255.255.255.0
IPV4 GATEWAY	Insert your network's gateway. Default 192.168.0.1.
NTP SERVER / TIMEZONE	Choose the nearest NTP Server (default „europe.pool.ntp.org") and your local time-zone (default „Europe/Rome").

Confirm your changes with the „Save Settings" button to store them. In order to be applied, the DIVUS-VS needs to be rebooted. Choose „Reboot System" from the dropdown menu, then the checkbox „Confirm" and finally push the „Reboot/Restart" button; the device will then be rebooted.

3.3 TROUBLESHOOTING

The following table contains typical messages for the status of single SIP clients – as shown in the section „SIP Peer Status" of the STATUS page.

MESSAGE	MEANING
UNKNOWN	The SIP client has not connected to the DIVUS-VS since the last boot. Check the connection and settings of the SIP clients if it is expected to connect. The pre-set accounts show this status as long as they are not in use.
OK	This SIP client has successfully registered at the DIVUS-VS server and can be called.
UNREACHABLE	This SIP client was registered but did not answer to the cyclical query from the server in a timely manner (or at all). An unreachable device cannot be called (although the server might still have its IP address stored). Check the connection between SIP client and server and the state of the SIP client.

The following table contains typical messages which the DIVUS-VS writes to its log file and the respective meaning:

MESSAGE	MEANING
Peer '101' is now REACHABLE/UN-REACHABLE! (number may vary)	SIP client „101" answered (or not) within the allowed time frame to the servers cyclical query. Only the queries which change the previous state are shown in the log file. An unreachable device cannot be called, but the server will continue to try to contact it. When the device responds again, the state will change to „reachable" and the according log entry will be generated. The message „unreachable" for a SIP client is often a sign of network latencies or other difficulties of the device to react to the server's inputs quickly. Check the network (e.g. WIFI, DNS, Gateway ...) and the related settings of the SIP clients and of the DIVUS-VS.
Registration from '"123"<sip:123@192.168.0.170>' failed for '192.168.0.42:10956' - Wrong password	SIP client „123" with IP address 192.168.0.42 failed to register at the DIVUS-VS – using a wrong password. Check the inserted data on the SIP client for any errors (also blank spaces).

`EXTENSION 123456 not found in context default …`	The call number „`123456`" could not be found in the context „`default`". Messages of this sort mean that a number was called which does not exist in the DIVUS-VS configuration. Check whether the number is wrong or whether it actually exists in the „`extensions_custom.conf`" file.
`Call from peer '101' rejected due to usage limit of 1`	The single SIP accounts of the DIVUS-VS have a limitation of one active channel at a time. This means the no SIP client is allowed to have more than one communication at the same time. In rare cases it might happen that an ongoing communication, and therefore an active channel remains opened i.e. is not closed correctly. In such a case the shown message will appear if the SIP client tries to open a new communication while still having the open channel. This may happen when the SIP protocol is not able to terminate the communication correctly – possibly due to network issues. Check the active channels on the status page while no communication is ongoing: it should be 0 in that case. If there should be open channels still, restarting the SIP service will solve the issue. On the EDIT page choose „Restart Asterisk" from the dropdown menu, set the checkbox „confirm" and then push „Reload/Restart". After the restart of the VoIP service there should be no active channels left.

3.4   EDITING THE CONFIGURATION FILES

3.4.1   INTRODUCTION

The configuration of the DIVUS-VS is done through so called configuration files. These are simple text files which may be opened and edited directly in the EDIT page.

This allows to make and apply changes in a fast way – without the need of an external editor just using the browser.

In order to apply the changes the service must be reloaded („Reload Asterisk") or in some cases also restarted („Restart Asterisk").

Warning: It is highly recommended to only edit the following files and leave all other files unchanged to grant the full functionality of DIVUS-VS:

- `sip_custom.conf`

- `extensions_custom.conf`

- `sip.conf`

Changes to any other configuration file should only be applied by DIVUS Support personnel.

3.4.2   SIP ACCOUNTS

The DIVUS-VS comes equipped with a set of pre-defined SIP accounts for internal units (101,102 …) and external units (901, 902, …). If other accounts should be needed, proceed as follows:

- Click the EDIT tab to open the corresponding page

- In the alphabetically sorted List choose „sip_custom.conf". The file's content will be shown in the lower part of the page, showing all the predefined and currently used accounts.

- Copy an existing entry starting from the header in square brackets down to its last parameter e.g.:

```
[101](internal)
defaultuser=101
secret=101
callerid=101<101>
```

- Paste the text at the end of the file and change the account's numbering. Please note that no double account is allowed! In this example the new entry for e.g. the new SIP account 179 would look like this:

```
[179](internal)
defaultuser=179
secret=179
callerid=179<179>
```

- Press the „Save Changes" button to store the changes in the configuration file.

- To have the changes applied you then need to reload the VoIP service. Choose „Reload/Restart" from the dropdown menu, set the „Confirm" checkbox and push the „Reload/Restart" button to reload the VoIP service.
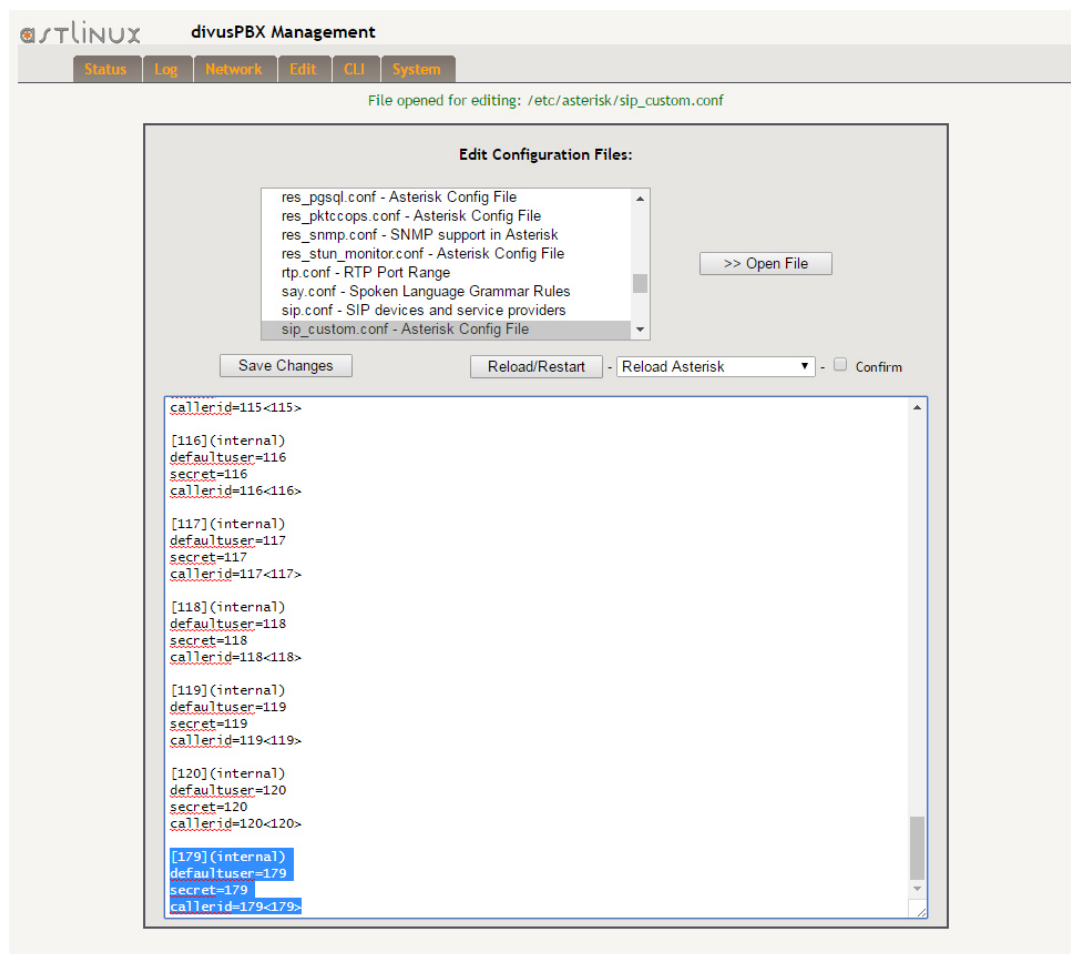
The same procedure can be used to edit existing SIP accounts. This is necessary only when you have to integrate third party devices with special needs.

Note: the file „`sip_custom.conf`" uses the syntax for templates. The templates „[external]" and „[internal]" can be seen at the beginning of the file. The single accounts which use these accounts can be recognized by the template name in brackets after the entry's name e.g. „`[901](external)`" represents the account „901", which uses the template „external". If you want to use parameters which differ from those of the templates e.g. to integrate a third party device needing a special configuration, you are free to use the default syntax (just remove the bracket so it will not inherit the template's settings; if needed, you may get in touch with the DIVUS support for special case settings.

3.4.3 DIRECT CALLS AND GROUP CALLS

For a SIP client to be callable, a corresponding call number must be set. This is done in the „`extensions_custom.conf`" file. In the default configuration of the DIVUS-VS there are call numbers for each of the predefined accounts, which is defined using the same numbering used for the accounts. So e.g. the SIP account „101" is pre-configured with the call number „101". If another SIP client calls this number, the SIP client „101" will ring. In this way each registered client can be called directly. Here is an example of such a configuration entry:

```
exten=>101,1,Dial(SIP/101,${VSRINGTIMEOUT},g)
```

If you add a new SIP account to the „sip_custom.conf" configuration file, the corresponding SIP client can register successfully but cannot be called yet, as the configuration is still missing a call number which addresses it. To set up the system to allow direct calls to the new account, proceed as follows:

- Click on the EDIT tab to open the corresponding page

- In the alphabetically sorted list choose „`extensions_custom.conf`". This will open the file in the lower part of the page, showing all the configured call numbers.

- Copy an existing (simple) entry e.g.:

```
exten=>101,1,Dial(SIP/101,${VSRINGTIMEOUT},g)
```

- Insert the text at the end of the file and change the numbers according to your new account. In this example, the entry for the new account "179" should look like this:

```
exten=>179,1,Dial(SIP/179,${VSRINGTIMEOUT},g)
```

- Click the „Save Changes" button to store the changes back to the file

- To have the changes applied you then need to reload the VoIP service. Choose „Reload/Restart" from the dropdown menu, set the „Confirm" checkbox and push the „Reload/Restart" button to reload the VoIP service.

Group calls are also set in the same configuration file. If you want to have a group call for calling only a subset of SIP clients, you can edit the file following these instructions:

- Click on the EDIT tab to open the corresponding page

- In the alphabetically sorted list choose „`extensions_custom.conf`". This will open the file in the lower part of the page, showing all the configured call numbers.

- Go to the group call you want to edit e.g.:

```
exten => 20060101,1,Dial( SIP/100&SIP/101&SIP/102&SIP/103&SIP/104&SIP/
105&SIP/106&SIP/107&SIP/108&SIP/109&SIP/110&SIP/111&SIP/112&SIP/113&SIP/114&
SIP/115&SIP/116&SIP/117&SIP/118&SIP/119&SIP/120,${VSRINGTIMEOUT},g)

exten => 20060101,2,Hangup
```

- Remove all the SIP clients from the "&"-chain which should not be called by the group call; each SIP client is represented with the syntax „SIP/<SIP-Account>" and separated from the others by the "&" symbol. Therefore, if you e.g. want to call the 3 SIP clients „101", „102" and „103" the edited result would look like this:

```
exten => 20060101,1,Dial( SIP/101&SIP/102&SIP/103,${VSRINGTIMEOUT},g)

exten => 20060101,2,Hangup
```

- Press the „Save Changes" button to store the changes in the configuration file.

- To have the changes applied you finally need to reload the VoIP service. Choose „Reload/Restart" from the dropdown menu, set the „Confirm" checkbox and push the „Reload/Restart" button to reload the VoIP service.

If you need to add additional group calls, proceed as follows:

- Click on the EDIT tab to open the corresponding page

- In the alphabetically sorted list choose „`extensions_custom.conf`". This will open the file in the lower part of the page, showing all the configured call numbers.

- Copy an existing group call entry e.g.:

```
exten => 20060101,1,Dial( SIP/100&SIP/101&SIP/102&SIP/103&SIP/104&SIP/
105&SIP/106&SIP/107&SIP/108&SIP/109&SIP/110&SIP/111&SIP/112&SIP/113&SIP/114&
SIP/115&SIP/116&SIP/117&SIP/118&SIP/119&SIP/120,${VSRINGTIMEOUT},g)

exten => 20060101,2,Hangup
```

- Insert the text at the end of the file. Edit the entry according to your needs, defining the number to be called and the callees, e.g.:

```
exten => 30056,1,Dial( SIP/101&SIP/102&SIP/103,${VSRINGTIMEOUT},g)

exten => 30056,2,Hangup
```

- Press the „Save Changes" button to store the changes in the configuration file.

- To have the changes applied you finally need to reload the VoIP service. Choose „Reload/Restart" from the dropdown menu, set the „Confirm" checkbox and push the „Reload/Restart" button to reload the VoIP service.

The single call numbers are divided into sections called "contexts" in the file „`extensions_custom.conf`". In the „sip_custom.conf" configuration file a context is assigned to each SIP account; a SIP client which registers using such an account, has only access to the given context.

In the default configuration all contexts are bound together, which allows anyone to call all other clients. If necessary, this behaviour may be changed, creating separate contexts for separate groups of clients such that they cannot reach each other. For such project specific needs we recommend to get in touch with the DIVUS support which will assist you in the custom configuration.

4   Additional functions

4.1   FLOOR CALLS PER UDP

DIVUS-VS allows to start floor calls using UDP telegrams. The server will listen on UDP port 8888 and start the floor call according to the received command.

The UDP command syntax for floor calls is the following:

```
CALL=<CHANNEL>,<ID>,<TIMEOUT>,<CONTEXT>,<EXTENSION>
```

The parameters' meaning is explained here:

- **CHANNEL**: Used to define the channel to be used. Usually the local channel is used, indicated through the string „Local/" followed by the call number to be used, e.g. „Local/20060101" if the call number „20060101", which will be defined in „`extension_custom.conf`" as group call, is to be used.

- **ID**: The caller ID for the floor call is set here using the format „XXX<XXX>". If e.g. „901<901>" is used, this number will appear as caller on the callees' SIP clients and will there be managed according to the clients' configuration. Using the DIVUS VIDEOPHONE software this ID will be used to couple the call to the configured video source (if any).

- **TIMEOUT**: This value decides for how long the call should last if no one should answer. Once reached the given number of seconds, the call will be ended.

- **CONTEXT, EXTENSION**: Context and number, which should be set when the call is answered. This allows e.g. to have a feedback sound. The following example will play a welcome sound/message:

```
[floorcall]
exten => fcAnswer,1,Answer()
exten => fcAnswer,2,Wait(2)
exten => fcAnswer,3,Playback(welcome)
exten => fcAnswer,4,Hangup
```

The following example starts the group call „20060101" and plays back the sound when one of the clients answers:

```
CALL=Local/20060101,901<901>,60,floorcall,fcAnswer
```

The context „[floorcall]" is part of the „`extensions_custom.conf`" configuration since version 2.0 and can therefore be used without further interventions. The same goes for the sound feedback using the call number (extension) „fcAnswer" - as shown above.