



Intercom Module Manual

Version 2.0
REV03-20161019

GENERAL INFORMATION

DIVUS GmbH
 Pillhof 51
 I-39057 Eppan (BZ) - Italy

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

CONVENTIONS

[KEY]	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.
	<i>Caution!</i> A dangerous situation may arise that may cause damage to material
	<i>Note</i> Hints and additional notes
	<i>New</i> New features

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1 Introduction

1.1 PREMISE

The Divus KNXCONTROL devices are able to handle numerous services of audio/video communication through a network using the VoIP (Voice over IP) technology, allowing the communication between different devices, like:

- IP phones and IP video phones
- IP based external units (door stations)
- APPLE or ANDROID tablets and smartphones
- External phone lines¹
- PC / MAC accessing OPTIMA through a browser (only KNXSERVER)

So basically any device used for the supervision can also communicate in bi-directional way with the other devices, manage intercom calls (also with video) or call internal and external phone numbers.

The VoIP communication services present in OPTIMA use the SIP protocol: every single hardware or software device that's to be used in your intercom system must therefore be compliant to this technology.²

1.2 REQUIREMENTS AND LIMITS

In order to use the VoIP functions offered by OPTIMA, you will need:

- A KNXCONTROL device with active INTERCOM license³
- At least one compatible client device or VoIP participant, as explained more in detail in the following chapters of this manual

The VoIP support of KNXCONTROL devices has been designed to extend the functionalities of the home automation system and should therefore not be seen as a professional telephony system, especially in the tertiary sector.

¹ In order to handle classic phone lines (like PSTN or ISDN), a corresponding VoIP gateway is needed (not sold by DIVUS)

² Please keep in mind that not all VoIP based applications – like e.g. SKYPE – use the SIP protocol

³ Please notice: the license depends on the version of your KNXCONTROL device; if you are using a KNX-M-SERVER, the INTERCOM license is NOT part of the product if not acquired separately. For all other server versions and KNXSUPERIO, a FREE INTERCOM license is pre-installed, with the limitations specified in the table. In order to be able to use ALL intercom functionalities explained in this manual, the FULL INTERCOM license is needed.

Particularly, the use of the VoIP system of KNXSERVER and KNXSUPERIO is not recommended above the following limitations:

FREE VERSION (INCLUDED WITH KNX-H/-A/-E SERVER)	FREE VERSION (INCLUDED WITH KNXSUPERIO)	FULL VERSION (NEEDS TO BE ACQUIRED SEPARATELY)
Only 1 external intercom unit	Only 1 external intercom unit	Up to 10 external intercom units
Up to 3 internal units (clients, mobile devices, IP phones, ...)	Up to 3 internal units (mobile devices, IP phones, ...) Note: the object types "USERS" and "CLIENTS" supported by KNX- SERVER, can't be used for intercom functionality on KNXSUPERIOS.	Up to 20 internal units (clients, mobile devices, IP phones, ...)
Max. 1 call group		No limit for configurable call groups
No external phone lines		No limit for configurable phone lines
15 concurrent calls		

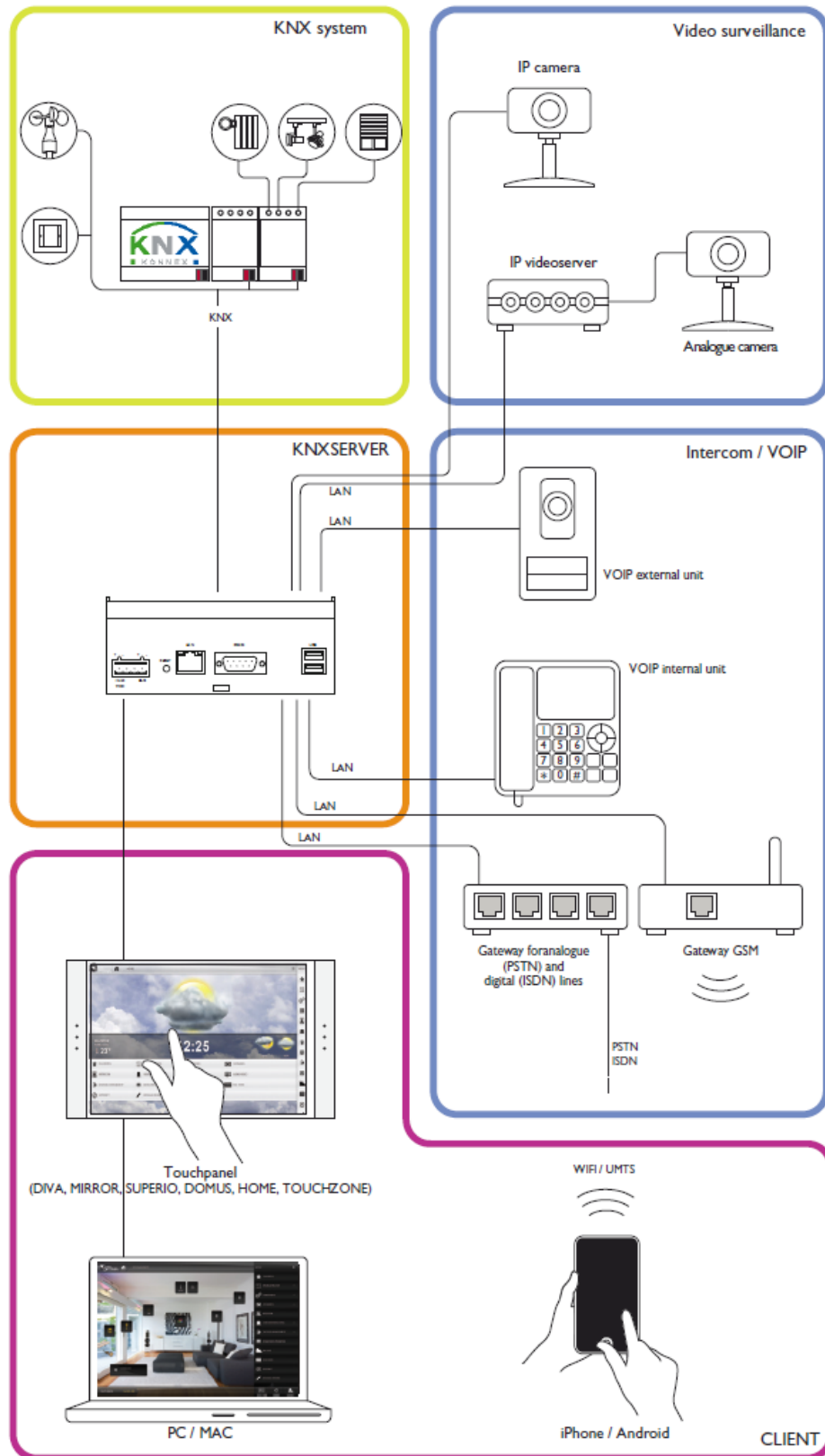
If those limits need to be exceeded (or if the intercom management needs to be done in tertiary or professional sectors), it is recommended to use a separate VoIP server which takes over the VoIP services from the KNXCONTROL device.



Warning: in contrast to KNXSERVER, KNXSUPERIO must perform a registration also locally, since this product – in opposition to the server – permits to access and interact with the VISUALISATION also locally. Therefore, a special client object with IP address 127.0.0.1 (internal loopback address) is used, which can be assigned to the desired VoIP number and accomplish the registration.

1.3 GENERAL STRUCTURE

The scheme on the following page shows a general function diagram of the VoIP management system (in this case shown in combination with KNXSERVER):



Once correctly configured and connected to the network, your KNXCONTROL device takes the role of "coordinator" of the single VoIP devices present in the system; everything happens using the existing network infrastructure, without the necessity of additional transmitter devices or particular network configurations. The

communication happens on the network (also WI-FI) with a very limited bandwidth occupation (provided that the previously listed limits are respected), so that there are no interferences with the other network functionalities.

As explained more in detail in the following chapters, the configuration of every single VoIP participant generally will require the following steps:

- The creation of a special VoIP object within the KNXCONTROL device (the needed object type depends on the type of VoIP participant you want to create) and the correct compilation of the required parameters
- In case of IP phones (hard- and software), mobile devices, intercom units or gateways, additionally the configuration of the device itself has to be done, in order to make it communicate with the KNXCONTROL device. Generally it is required that you specify the IP address of your KNXCONTROL product as VoIP server address within the configuration of the device, and then specify the username and password that you previously created in the configuration area of the OPTIMA interface.

1.4 EXAMPLE CONFIGURATION

The following example should help you to understand the new types of objects and therefore help you to find the correct object type for your system.



Hint: In this example a KNXSERVER is used. Please remember that when using a KNXSUPERIO, only mobile devices, DIVUS TZ, IP phones and external units can be used but not other PCs / panels acting as clients. Furthermore, the client object of the KNXSUPERIO itself (127.0.0.1, check out the hint on page 6) must be connected to a VoIP number too in order to work as expected.

Let's assume we have a system with the following components and IP addresses:

DEVICE	IP ADDRESS
KNXSERVER	192.168.0.110
DIVUS SUPERIO 10"	192.168.0.101
DIVUS SUPERIO 15"	192.168.0.102
DIVUS TOUCHZONE	DHCP
Hardware IP Phone	DHCP
Mobile device	DHCP (WI-FI)
Intercom – door station	192.168.0.120

All of the devices (except the IP phone and the door station) will access the VISUALISATION of KNXSERVER. Furthermore, on a call of the door station all client devices should start to ring.

For the configuration of the door station it is necessary to create an EXTERNAL UNIT object and afterwards the door station needs to register using the data specified in the EXTERNAL UNIT object. Please find the information about the configuration of EXTERNAL UNITS in chapter 4 of this manual. We will just assume that the door station will dial the VoIP NUMBER 200 when the door bell is ringed.

Now let's refer to the different client devices: first you will have to understand which object types you will have to use for the different client devices. As explained in the box on the previous page, two of the three available object types require a browser with JAVA support. This means, we will first check which devices offer this support:

DEVICE	BROWSER WITH JAVA SUPPORT
DIVUS SUPERIO 10"	Yes
DIVUS SUPERIO 15"	Yes
DIVUS TOUCHZONE	No
Hardware IP Phone	No
Mobile device	No

We can see that only the *Windows*-based devices (Touchpanels of the DIVUS SUPERIO line) offer the required JAVA support. Consider this as a rough guide: all PC-based devices (Windows / Linux / MacOS based notebooks, PCs, Touchpanels, etc.) generally fulfill the JAVA compatibility, while hardware VoIP devices (like IP phones, IP video phones etc.), softphones or mobile devices (Android, iOS, etc.) do NOT offer this functionality.

1.4.1 VOIP – PARTICIPANTS

The DIVUS TOUCHZONE, the IP phone and the mobile device respectively need an object of type VoIP PARTICIPANT to be configured according to the information in chapter 3.

Once the 3 objects have been created on the KNXSERVER (Optima), the devices still have to register on the server; The DIVUS TOUCHZONE now could use the integrated VIDEOPHONE APP in order to register on the KNXSERVER, the IP phone could be configured with those parameters through its web interface and the mobile device could use the parameters in combination with one of the apps recommended in chapter 3.1.

As next step, we will take a closer look at the 2 DIVUS SUPERIOs; we have seen that both devices offer JAVA support, which means that we can use either the CLIENTS or the USERS objects.

1.4.2 CLIENTS

Let's first check out the CLIENTS; for every device accessing the KNXSERVER, such a CLIENT object will automatically be created; this means that in our example system, we will find the following client objects:

- DIVUS SUPERIO 10"
- DIVUS SUPERIO 15"
- DIVUS TOUCHZONE
- Mobile device

Please note that they will be listed by their IP address, not by those names; now you probably will wonder why there are listed also the DIVUS TOUCHZONE and the mobile device: this because the CLIENT objects are not limited to offer VoIP functionality, but also offer other functions like automatic PAGE JUMPS; so even if you can't use the VoIP functionality through the CLIENT object, you could still use it to perform page jumps on the mobile device and the DIVUS TOUCHZONE; information about the other usages of the CLIENT objects can be found inside the OPTIMA ADMINISTRATOR MANUAL.

Since we will focus on the VoIP functionality, we can ignore (or even delete) the CLIENT objects for the DIVUS TOUCHZONE and the mobile device; if we instead open the property pages of the other 2 client devices, we will see that there is a field for a VoIP NUMBER (max. 6 digits); if we assign the number 101 to the 10" SUPERIO and 102 to the 15" SUPERIO and save the settings, we have already completed the VoIP configuration of those 2 devices. Now it would immediately be possible to open the intercom pop-up on one of the 2 devices and to call

the other one using the assigned number. So, in contrast to the configuration via VoIP PARTICIPANTS, the CLIENTS need only a SINGLE configuration step and no further action on the SUPERIO devices. Details about the CLIENTS and the use of the intercom pop-up can be found in chapter 2.

1.4.3 USERS

The usage of the USERS is very similar to the one of the CLIENT object; even in the properties page of the USERS you will a field for a VoIP NUMBER (max. 6 digits), just as for the CLIENT objects. The differences are the following:

- CLIENTS are always limited to one device, through their IP address. So when assigning a number to a CLIENT, this number will always point to one single client device.
- USERS instead are not limited to a certain device; you can define which device should login using which user; this means that the VoIP configuration is more flexible and dynamic and can be changed more easily just by switching users. Furthermore, users can be logged in on more than one device and therefore, dialing the VoIP number of the user will cause all client devices that are currently logged in with that user to ring.
- CLIENTS are created automatically, when logging in, based on their IP address. So you can only configure them when the corresponding client is online or has connected at least once to the KNXSERVER.

So you can see that the users bring a few advantages under some circumstances; nevertheless, you are completely free to select between the two object types; it is even possible to use them AT THE SAME TIME: if for example we take the configuration from the last chapter (using the CLIENT objects) and now create a user "superio" with number "110" which will be used to login on both SUPERIO devices, we will have the following scenario:

- If the number "101" is dialed from another device, the SUPERIO 10" will ring
- If the number "102" is dialed from another device, the SUPERIO 15" will ring
- If the number "110" is dialed from another device, both SUPERIO panels will ring and the first one answering will take the call (meanwhile the other one will stop ringing)

So it may even make sense to use both CLIENTS and USERS at the same time! Further details on the configuration of the VoIP functionality using the USERS can be found in 6.

1.4.4 SETUP CONCLUSION

We have now seen an example for the three different object types for the creation of the INTERNAL UNITS.

In order to achieve the desired functionality, there is still one open point: we have seen how to configure the different devices, but we did not establish the call from the external unit to the client devices yet; we have just defined that - pressing the ring button - the door station should dial the number "200".

In order to make all client devices ring, we have to create an object of type CALL GROUP; in the properties page of this object we can set the VoIP NUMBER to "200" (in order to match with the number dialed by the external unit) and then we just have to add the single client devices as MEMBERS of the call group; when the CALL GROUP is dialed, it will automatically redirect the call to all objects in the MEMBERS section of the CALL GROUP.

In our example case this would be:

- The 3 VoIP PARTICIPANT objects for "DIVUS TOUCHZONE", "Hardware IP phone" and "Mobile device"
- And, depending on your selection:
 - EITHER the 2 CLIENT objects for the 2 SUPERIO panels

- OR 1 USER object (if both SUPERIO use the SAME user)
- OR 2 USER objects (if each SUPERIO has its own user)

In this way, in case of a call from the door station, all client devices will start to ring and the first one answering will take the conversation, stopping all other clients' ringtone.

Feel free to experiment with the different VoIP functionalities of the KNXCONTROL products, trying to configure your VoIP system in different ways, using the possibilities explained on the further pages of this manual.

2 Clients (KNXSERVER only)

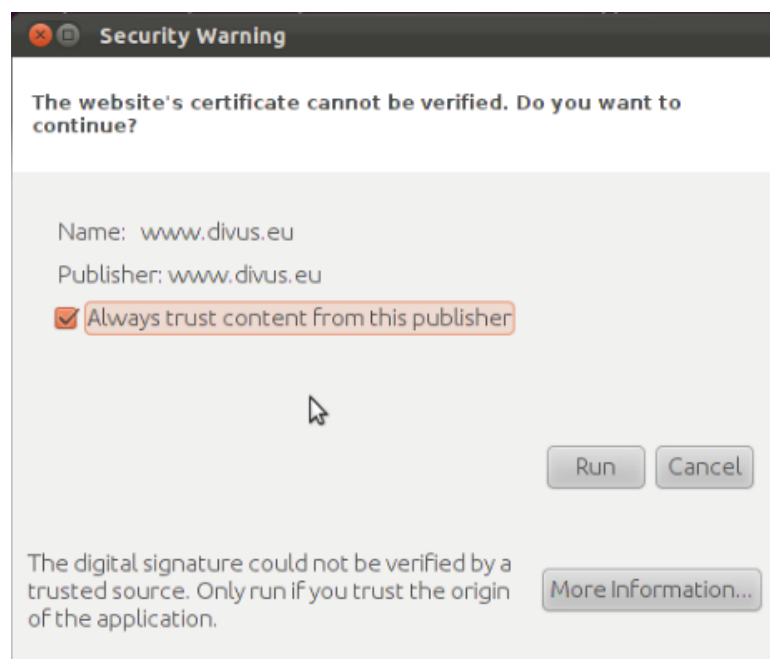
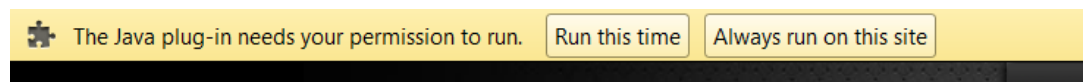
2.1 INTRODUCTION

Every computer endowed with a browser (provided that it is compatible with the JAVA technology) can be used for starting and receiving audio/video communications through the VoIP services of KNXSERVER. KNXSUPERIO will as well show the CLIENT objects, but only of the mobile devices (which can't use the CLIENT object for VoIP communication) and of itself (127.0.0.1 - where the VoIP number of KNXSUPERIO needs to be defined). Since KNXSUPERIO supports no Desktop PCs / Touchpanels, this chapter is of no interest when using a KNXSUPERIO.

2.2 CONFIGURATION

If the INTERCOM module is enabled, every JAVA compatible device accessing KNXSERVER through a browser is automatically registered to the VoIP services of KNXSERVER (without the necessity of any configuration).

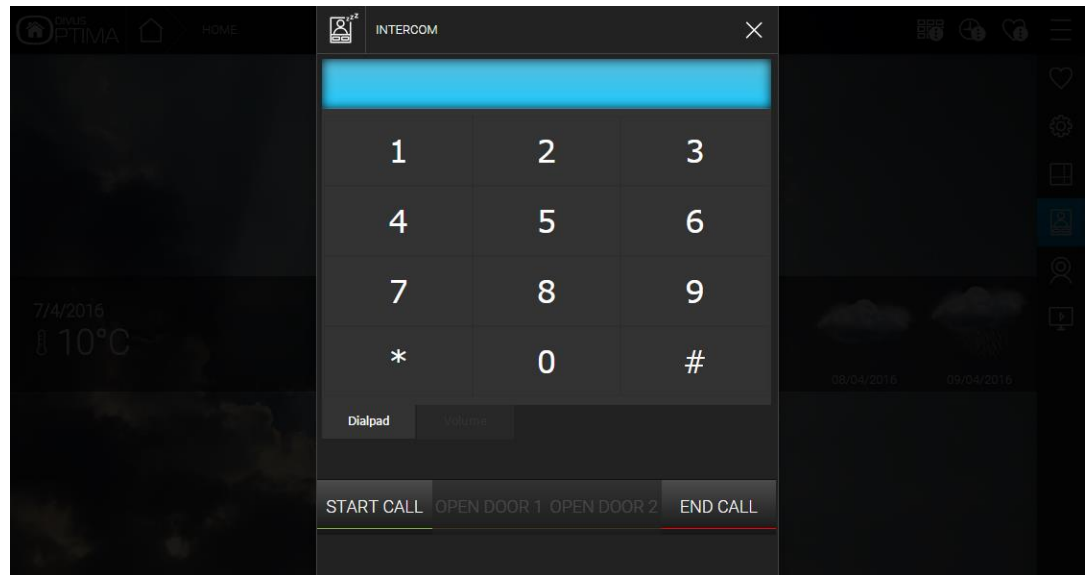
When accessing the interface for the first time, normally an authentication request will be shown in order to permit the usage of the VoIP functions within the browser (depending on your system, the message may appear in different ways, please find an example below):



If these alerts are ignored, using the VoIP service will not be possible. To avoid the alert appearing on every restart of the browser, it is recommended to set the checkbox making the choice durable.

2.3 USE OF THE VOIP FUNCTIONS

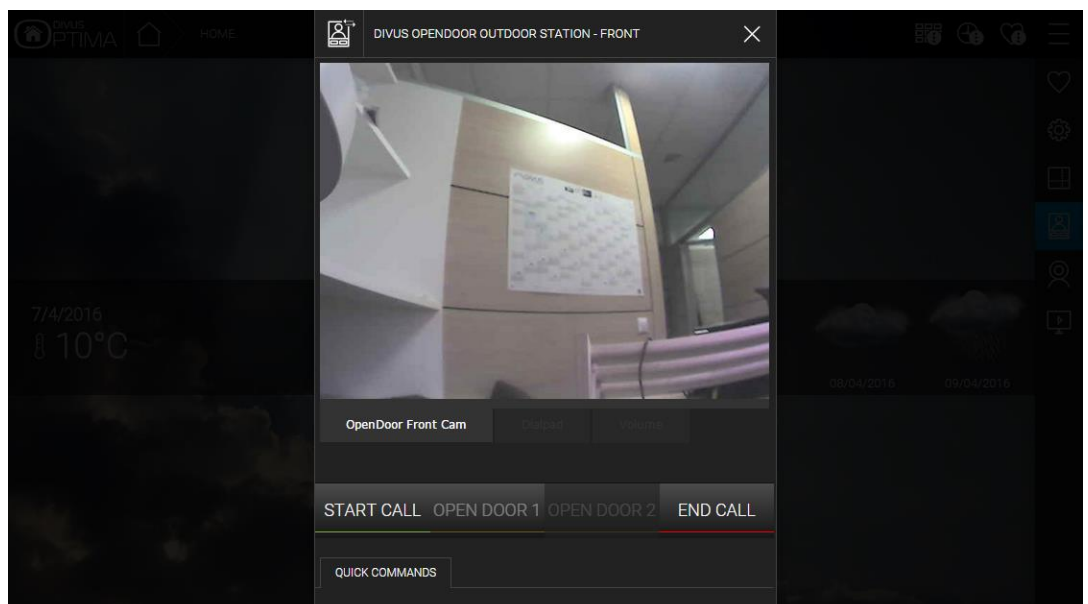
At this point it is already possible to start calls to other VoIP participants (if previously configured in Optima): just select the entry "INTERCOM" from the NAVIGATION MENU in the VISUALISATION, digit the number you want to dial and hit the button "START CALL":



In order to close the call or the pop-up window, just hit the "END CALL" button.

Please note that from this moment you are registered on the VoIP server through the CLIENT object as an ANONYMOUS participant, since you did not assign a VoIP NUMBER to the CLIENT object corresponding to the IP address of your device. This means that you are able to make outgoing calls, but you can't be reached by other participants. In order to allow this, please access the administration menu of Optima, open the properties page of the CLIENT object corresponding to the IP address of your device and insert a free VoIP NUMBER.

If you receive a call from another VoIP participant, the same pop-up window will be automatically opened and a ringtone will be enabled in order to notify about the incoming call; if for the calling VoIP device also a videosegment has been configured (as explained in the next chapters), it will be shown instead of the dial pad:



In case of an intercom unit, during the conversation also the button "Open door" is available to activate the opening of the corresponding gate (through the integrated relay of the intercom unit). Furthermore, it is always possible to visualise the dial pad by pressing the corresponding entry among the available "TABS"; in the same way it is possible to switch between the different cameras associated to the calling VoIP participant.

Further information about intercom units and their configuration can be found in the corresponding chapter within this manual.

3 VoIP Participants

3.1 INTRODUCTION

Your KNXCONTROL product is able to handle the following devices as VoIP participants and enable audio/video communication between them:

- Hardware IP phones or video phones, if compatible with the SIP standard; find below some IP phones already tested and working with the KNXCONTROL family:
 - Cordless SIEMENS GIGASET IP
 - GRANDSTREAM phones and video phones
 - SNOM phones
- Phone / VoIP software (so called "softphones"), if compatible with the SIP standard; some examples:
 - Linphone (Windows, Mac, Linux)
 - X-Lite (Windows, Mac, Linux)
 - Bria (Windows, Mac)
 - DIVUS VIDEOPHONE (Windows)
- Smartphones with native SIP support (e.g.: NOKIA, BLACKBERRY)
- Apps for mobile devices with APPLE IOS or ANDROID; some examples:
 - Linphone
 - Vippie
 - Bria
 - DIVUS VIDEOPHONE 4 app (Android, iOS)

This chapter will explain in detail how to configure a VoIP participant in your KNXCONTROL device and how to register it correctly to use the VoIP functionalities.

3.2 PREPARATION OF THE KNXCONTROL DEVICE

In order to enable a new VoIP participant, it is necessary to create a corresponding object within your KNXCONTROL device. In order to do so, please follow the steps below:

- Access the configuration area of the OPTIMA interface
- Select "TECHNOLOGIES" from the administration menu, then "INTERCOM" and finally "VoIP PARTICIPANTS"
- Please click on the "ADD" button within the toolbar at the bottom; a new object will be created and listed as "New VoIP participant"
- Now open the properties page of the new created object by clicking on the "EDIT" button in the toolbar (or alternatively on the "three dots" on the right side).



Hint: If the "INTERCOM" section is not present within the menu "TECHNOLOGIES" of your KNXCONTROL product, it means that you have not activated the INTERCOM license. In this case, please contact the support of DIVUS in order to get additional information on how to purchase the license.

The properties page of the new VoIP participant is similar to the one shown below:

The screenshot shows the DIVUS OPTIMA software interface. The top bar includes the logo, version (2.0.1), and buttons for SAVE, CLEAR CACHE, EXPERT, LOGOUT, and VISUALISATION. A search bar is on the left. The main navigation menu on the left lists various setup options, with 'New VoIP participant 2' selected under 'Intercom'. The main content area is titled 'New VoIP participant 2' and contains the following sections:

- Object properties**
 - General information**
 - Name:
 - Number:
 - Password:
 - Caller ID:
 - Buttons: **CLOSE** (red), **UPDATE CONFIG.** (green)
- Connected IP cameras** (table with columns: Name, Description / ETS name)
- Connected quick commands** (table with columns: Name, Label)
- Connected scenarios** (table with columns: Name, Action, Value, Order)
- Outgoing connection** (table with columns: ID, Name, Condition, Action, Value)
- Incoming connection** (table with columns: ID, Name, Condition, Action, Value)

NAME	Name, identifies the VoIP participant within the software
NUMBER	Number under which this VoIP participant registers and can be reached (max. 6 digits)
PASSWORD	Password to be set in the configuration of the VoIP participant, in order to complete the registration on the integrated VoIP server
CALLER ID	Optional label that will be displayed instead of the phone number on other devices, when they receive a call from this VoIP participant

The examples within this manual assume that the enumeration of the VoIP participants starts with 101 and is incremented for every new device (102,103,104,...), even if the enumeration can be freely selected, as long as there are no duplicate numbers.



Hint: In order to simplify the configuration of the VoIP participants, it is recommended to use the same value for the NUMBER and the PASSWORD.

Once all data has been set, please click on the "UPDATE CONFIG." button in order to save the configuration and enable the newly created VoIP participant. Now a VoIP participant can register on the KNXCONTROL device using the parameters previously configured, as explained in chapter 3.3.

3.2.1 QUICK COMMANDS

It is possible to connect one or more "QUICK COMMANDS" to the VoIP participant, simply by dragging the desired object from the search function into the corresponding section of the properties page.

These quick commands will be available within the intercom pop-up window in the VISUALISATION of the KNXCONTROL device and permit to have access to certain functions even during active calls. Further information about this topic can be found in chapter 4.2.3 dedicated to the external units, which typically use this feature.

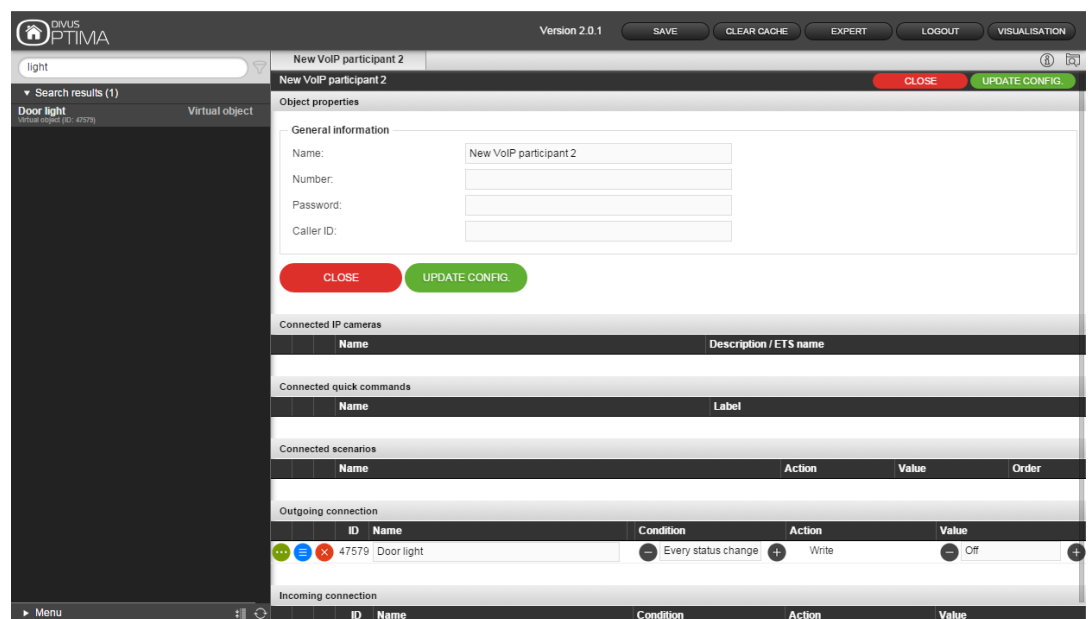
3.2.2 CAMERAS

It is possible to assign one or more camera signals to every VoIP participant, which will be shown in the intercom pop-up during calls; further information about this topic can be found in chapter 4.2.2 dedicated to the external units, which typically use this feature.

3.2.3 EVENTS

It is possible to connect one or more events to each VoIP participant; in this way, actions can be launched in dependency of incoming / outgoing calls from / to the selected VoIP participant.

In order to launch one or more events when the VoIP participant receives a call, it is necessary to define **OUTGOING CONNECTIONS**, using the corresponding section of the properties page:

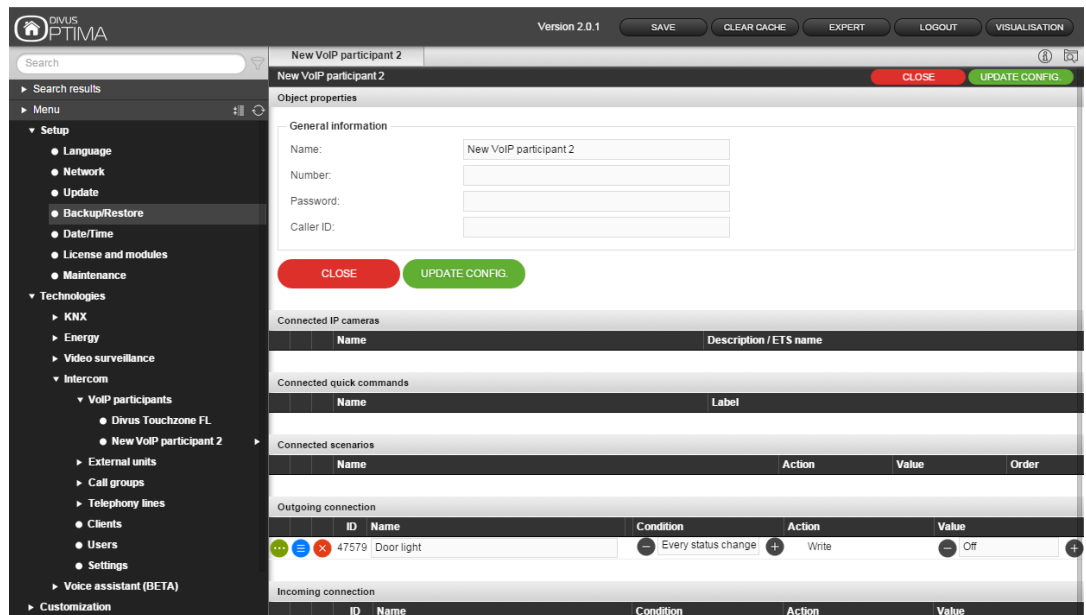


When dragging an object into this section (using the search function), it is possible to set the **CONDITION** of the event, by selecting between the following options:

- "If Wait..." → the event is launched when the call is closed and the VoIP participant returns into IDLE state, waiting for new calls
- "If Call in process" → the event is launched when the VoIP participant is called and starts ringing
- "Every status change" → the event is launched in both cases described above

Based on the selected object, it is also necessary to define the action to be executed and – if required – the value to be passed. Further information regarding the management of **EVENTS** in the OPTIMA interface can be found in the **ADMIN MANUAL** of your KNXCONTROL device.

If instead the VoIP participant should be called when a certain event occurs in the software, it is necessary to configure a **PASSIVE EVENT**: in this moment, the object that should trigger the call must be dragged in the corresponding section...



... and the CONDITION, which starts the call, must be set, selecting from the possible values of the object dragged in this section.

If the object assumes the state configured in the column CONDITION, the OPTIMA interface will start a call towards the VoIP participant currently in configuration. The VoIP participant will see an incoming call (the VoIP participant will show KNXSERVER or KNXSUPERIO as caller name of the incoming call).

As soon as the VoIP participant answers the incoming call, it will be put into communication from the OPTIMA interface with the phone number (could be another VoIP participant, a call group or a user) specified in the column "VALUE" of the PASSIVE EVENT: in this field you can...

- Either input the number of another VoIP participant configured in your KNXCONTROL device; In this case, the selected VoIP participant will be called.
- ...or input the number of a call group configured in your KNXCONTROL device (will be explained in a further chapter of this manual). In this case all the members of the call group will be called.
- ...or (in case of KNXSERVER) the name / number of a user. In this case all clients currently logged with the selected user will be called.

Instead of typing in the desired values, you can also drag the desired objects into the field in the column "VALUE". In this case it is necessary to transform the text field into a drop zone by clicking on the red button on the right side; afterwards, the desired object can be located using the search function and can be dragged into the corresponding field, as shown in the following screenshot (where the "admin" user has been connected using drag & drop):

The screenshot shows the DIVUS OPTIMA web interface. At the top, there's a navigation bar with 'Version 2.0.1', 'SAVE', 'CLEAR CACHE', 'EXPERT', 'LOGOUT', and 'VISUALISATION'. Below this is a search bar containing 'admi'. The left sidebar shows search results (7) including 'Administrators', 'Load control', 'Load alarm', 'Step', 'Max. limit', 'Min. limit', and 'admin'. The main content area is titled 'New VoIP participant 2' and contains the following sections:

- Object properties**: A section with 'General information' containing fields for Name (filled with 'New VoIP participant 2'), Number, Password, and Caller ID. Below these are 'CLOSE' and 'UPDATE CONFIG.' buttons.
- Connected IP cameras**: A table with columns 'Name' and 'Description / ETS name'.
- Connected quick commands**: A table with columns 'Name' and 'Label'.
- Connected scenarios**: A table with columns 'Name', 'Action', 'Value', and 'Order'.
- Outgoing connection**: A table with columns 'ID', 'Name', 'Condition', 'Action', and 'Value'. It contains one entry: ID 47579, Name 'Door light', Condition 'Every status change', Action 'Write', Value 'Off'.
- Incoming connection**: A table with columns 'ID', 'Name', 'Condition', 'Action', and 'Value'. It contains one entry: ID 47580, Name 'Call pushbutton', Condition 'If On', Action 'VoIP call', Value 'admin'.

3.3 IP (VIDEO-) PHONES / SOFTWARE / APPS

In order to use your IP (video-) phone, softphone or similar in combination with your KNXCONTROL device, first a corresponding VoIP participant object needs to be created, as described in chapter 3.2; afterwards, the data from the VoIP participant can be used to register the IP phone / softphone on the KNXCONTROL device.



Hint: Independent of the type of VoIP device you try to register (e.g. an IP phone, a PC with a VoIP software, a mobile device with a VoIP app), must support the SIP protocol. SKYPE for example does not use this technology and therefore can't be used in combination with the KNXCONTROL family.

Generally, it is necessary to follow the steps below:

- Use a browser to connect to the settings page of the IP phone (HARDWARE) OR open the settings page of your (mobile) softphone application (SOFTWARE)
- Create a new "SIP account" or "SIP profile" for your KNXCONTROL device, specifying:
 - The IP address of the KNXCONTROL device wherever the VoIP server reference is needed (or VoIP "domain", "proxy", "host" etc.)
 - The phone number of the corresponding VoIP participant wherever the "user ID" or similar is required
 - The password of the corresponding VoIP participant wherever the "auth ID" or similar is required
 - Optionally the label used to identify the IP phone / softphone wherever the "caller ID" or similar is required

At the end of the configuration (after rebooting the IP phone / softphone, if required), the registration on the KNXCONTROL device should take place and therefore the starting and receiving of calls should be possible. Generally, if the registration was not successful, great part of the IP phones / softphones will allow the check of the registration state and give information about eventual errors (like wrong IP, incorrect user or password data etc.)

Unfortunately the configuration menus of the single IP phones, softphones and mobile apps can be very different; therefore there is no general documentation that could be provided for sample purposes within this manual. If you are using either the DIVUS VIDEOPHONE software or app, you can find further information in the corresponding manual present on the DIVUS homepage (www.divus.eu). Otherwise, please refer to the

documentation of the used IP phone / soft phone. In chapter 3.1 you can find some tested devices / applications for your reference.

4 External units

4.1 INTRODUCTION

In addition to the previously seen VoIP participants it is also possible to integrate into your KNXCONTROL product one or more IP based EXTERNAL UNITS (also with VIDEO), if compatible with the SIP protocol. After the configuration of the corresponding objects, it is possible to receive calls from the external units in the same way as between VoIP participants, with the possibility to show in real time the video signal of one or more cameras, to open the door or to execute other actions (like turning the outdoor lights on).

4.2 PREPARATION OF THE KNXCONTROL DEVICE

4.2.1 AUDIO CONFIGURATION

As already seen for the IP phones / softphones, also for the external units it is necessary to create an object within the OPTIMA interface for every single unit that should access the VoIP services.

Therefore, first of all the entry "EXTERNAL UNITS" in the section "INTERCOM" under "TECHNOLOGIES" must be selected and a new object must be created using the "ADD" button within the TOOLBAR on the left bottom.

Then please open the properties page of the new object, which should appear similar to this:

The screenshot shows the DIVUS OPTIMA web interface. The top navigation bar includes 'SAVE', 'CLEAR CACHE', 'EXPERT', 'LOGOUT', and 'VISUALISATION'. The left sidebar contains a menu with categories like 'Menu', 'Setup', 'Technologies', 'KNX', 'Energy', 'Video surveillance', 'Intercom', 'VoIP participants', 'External units', 'Call groups', 'Telephony lines', 'Clients', 'Users', 'Settings', 'Voice assistant (BETA)', 'Customization', 'Visualisation', 'Advanced functions', 'Data-Logging', and 'Users and permissions'. The main content area is titled 'Divus Opendoor Outdoor station - Front' and shows the 'Object properties' configuration page. The 'General information' section includes fields for Name, Number, Password, DTMF for call start, DTMF for dooropener, DTMF for dooropener (2), DTMF for call end, and Delay when sending multiple DTMF tones. Below this are sections for 'Connected IP cameras', 'Connected quick commands', and 'Connected scenarios'. The 'Connected IP cameras' section shows a table with columns for Name and Description / ETS name, containing one entry: 'OpenDoor Front Cam'. The 'Connected quick commands' section shows a table with columns for Name and Label, containing one entry: 'Door light' with label 'Light'. The 'Connected scenarios' section shows a table with columns for Name, Action, Value, and Order.

The following parameters are available:

NAME	Identifies the external unit within the OPTIMA interface.
NUMBER	Number under which the external unit registers and can be reached (max. 6 digits).
PASSWORD	Password to be set in the configuration of the external unit in order to complete the registration on the integrated VoIP server.
ID NAME	Optional label that will be displayed instead of the phone number on other devices, when they receive a call from this external unit.
DTMF-TONES	If the external unit needs one or more DTMF tones after the ACCEPT signal in order to start the conversation, please specify them here.
DOOROPENER SIGNAL DOOROPENER SIGNAL (2)	Please use these fields to specify the DMTF tones which activate the primary and secondary (if present) door opener relay of the external unit.
DTMF TONE TO CLOSE CONVERSATION	If the external unit needs one or more DTMF tones before the HANGUP signal in order to close the conversation, please specify them here.
DELAY WHEN SENDING MULTIPLE DTMF	If the configured DTMF tones consist of more than one single tone, it is possible to specify the interval time (in milliseconds) between one and another signal. This because some external units will not accept tones if they are sent out too quickly.

Once the configuration has been updated, please hit the button "UPDATE CONFIG."

4.2.2 CAMERAS

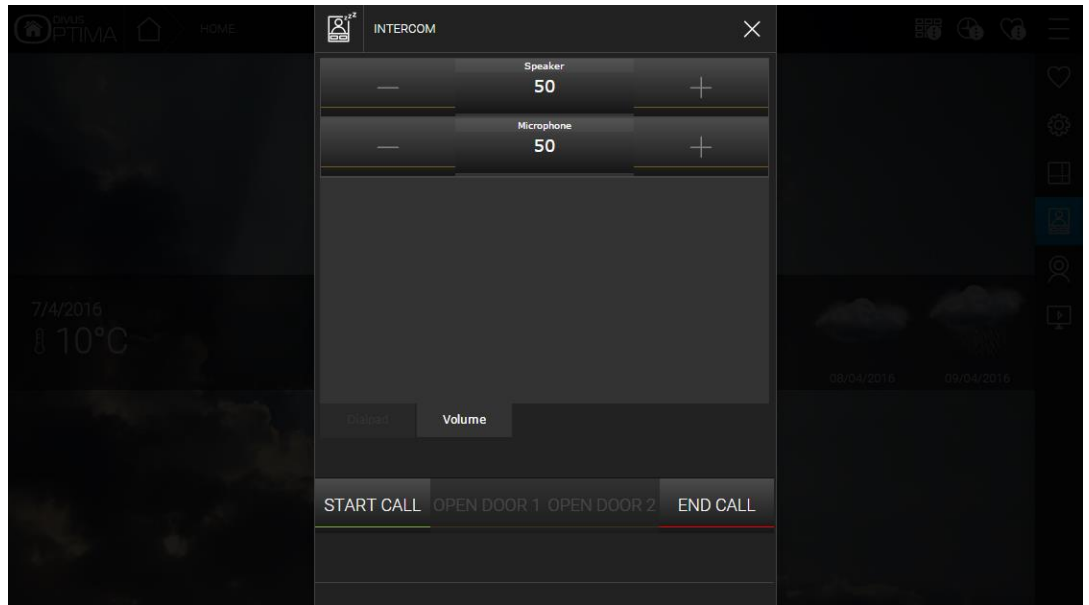
Once the configuration of the VoIP / audio part of the external unit is completed, you can connect one or more cameras to the external unit. Therefore, in addition to the video signal of the external unit itself, it is also possible to add other camera signals, even if completely independent from the door station and to show them in the intercom pop-up.

In order to connect the cameras to the external unit, they just need to be located through the search function and can then be dragged into the section "CONNECTED IP CAMERAS" (see last screenshot).

This means that the video signal of the external unit itself must be configured as a camera object, just like every other camera signal, too. Further details on the configuration and integration of video signals can be found within the OPTIMA ADMIN MANUAL of your KNXCONTROL device.

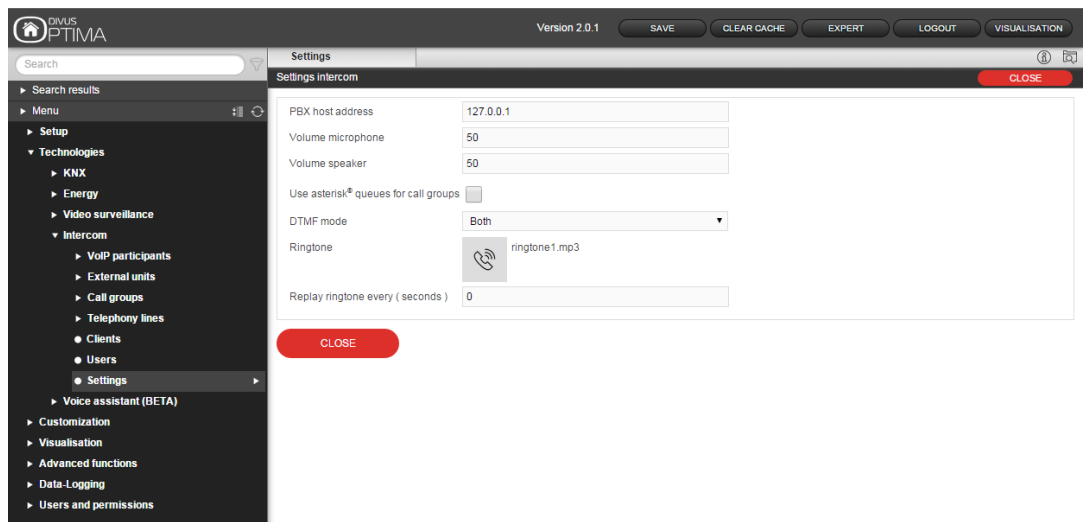
4.2.3 AUDIO CONTROLS

Beside the TABS for switching between the different cameras, the intercom pop-up also contains a TAB named „VOLUME“, which offers settings to control loudspeaker and microphone also during a call. The currently set volume of the speaker and the currently set sensitivity of the microphone are shown in the corresponding controls. The values can be changed through the corresponding "+ / -" buttons; this adaption of the volumes is also possible during a running intercom conversation. While "0%" corresponds to muting the audio signal, "100%" corresponds to the currently configured system volume:



Due to the different audio levels, which can occur when using different VoIP devices for communication, these values are only temporary adjustments that are lost when restarting the device. In the configuration area of the respective KNXCONTROL device, also default values for the speaker volume and the sensitivity of the microphone can be set and stored permanently; The KNXCONTROL device will load these values after every reboot as default values.

The corresponding settings are available under „TECHNOLOGIES → INTERCOM → SETTINGS“:



The audio settings must be adapted to fit with the installed door station. It is recommended to start with low volumes and then increase them gradually until reaching an optimal voice quality. In addition, in this menu you can set the IP address of your VoIP server; this setting called "PBX host address" is required only when an external VoIP server is used.

PBX-HOST ADDRESS	IP address of the VoIP server to be used. Normally the KNXCONTROL device itself is used as VoIP server („127.0.0.1“). If an external VoIP server is used (e.g. in condominiums, apartment buildings), the IP address of the used VoIP server must be inserted here.
VOLUME MICROPHONE	Default setting for the microphone’s sensitivity of the used KNXSUPERIO / the used client devices in percent.
VOLUME LOUDSPEAKER	Default setting for the loudspeakers volume of the used KNXSUPERIO / the used client devices in percent.



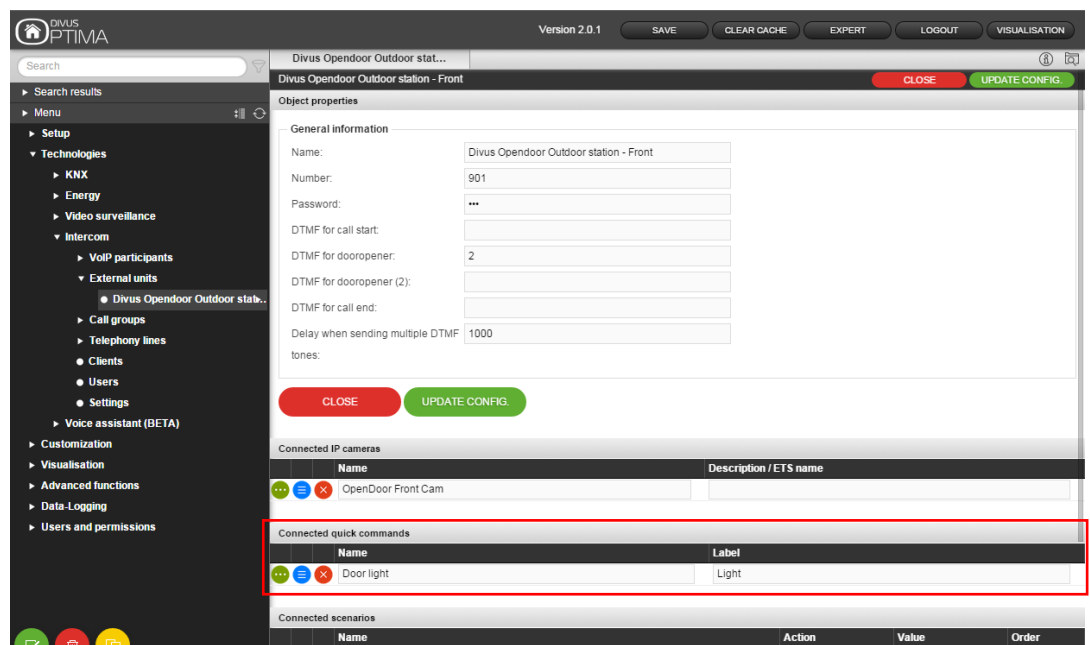
Attention: please note that in case of excessive sensitivity of the microphone and high volumes for the loudspeaker, interruptions and echo can occur in the communication, since the built-in AEC (Acoustic Echo Cancellation) cannot process distorted signals!



Hint: the intercom volume control affects only ongoing voice communication (e.g. door communication). Any acoustic signals, including the ringtone for intercom calls, are reproduced according to the configured system volume, independently from the volumes configured in the intercom pop-up. Furthermore, please note that this volume control is hardware-dependent: in case of KNXSUPERIO you will directly control the intercom volume of KNXSUPERIO; in case of KNXSERVER instead you will only control the intercom volume on the client device where you are actually changing the values.

4.2.4 QUICK COMMANDS

It is possible to add one or more objects as “QUICK COMMANDS”, simply by dragging them from the search function into the corresponding section of the properties page of the external unit. Those objects consequently will be accessible directly from within the intercom pop-up, as shown in the following screenshot (object “Door light”):



4.3 EXTERNAL UNITS

Once the object for the external unit has been created within your KNXCONTROL product, it is necessary to configure the door station itself in order to register with the given data and to use the VoIP services.

Generally, the configuration is very similar to the one of IP phones; even in this case you will have to access the IP address of the door station through a browser and configure the following parameters in the configuration page of the external unit:

- The IP address of the KNXCONTROL device wherever the VoIP server reference is needed (or VoIP "domain", "proxy", "host" etc.)
- The phone number of the corresponding external unit object wherever the "user ID" or "user name" is required
- The password of the corresponding external unit object wherever the "auth ID" or "password" is required

Even here, the configuration interfaces may vary from door station to door station. If using a DIVUS product (like OPENDOOR or the VIDEOPHONE-BOX), you will find the necessary documentation within the manual of the corresponding product.

5 Call groups

5.1 INTRODUCTION

It is possible to create one or more CALL GROUPS within your KNXCONTROL product: these objects permit to redirect VoIP calls to more than one target number: the first member of the CALL GROUP that answers the call will take the conversation, terminating the call on all other members.

5.2 CREATION OF A CALL GROUP

In order to create a new call group, please select the corresponding entry from the configuration area, under "TECHNOLOGIES" → "INTERCOM"; by clicking on the "ADD" button in the TOOLBAR at the bottom you can create a new call group. Accessing to its properties page, you will be able to configure the following parameters:

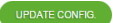
NAME	Identifies the call group within the OPTIMA interface
NUMBER	Number (max. 6 digits) under which the call group can be reached (dials all connected members)
QUEUE ID	Keyword (without spaces or special characters) used by the system to identify the call group
TIMEOUT	Timeout after which the call is terminated automatically if no one of the members accepts it

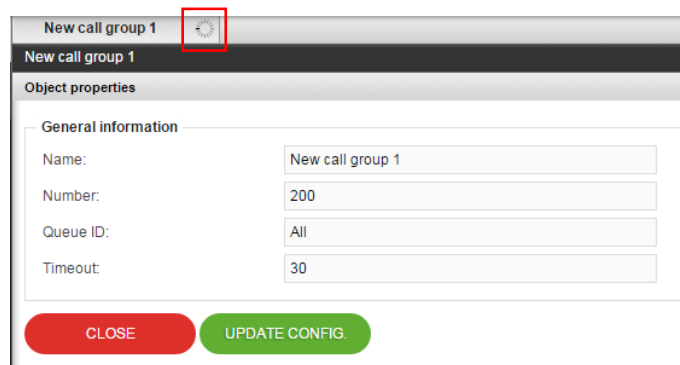
The screenshot shows the DIVUS OPTIMA configuration interface. The top navigation bar includes buttons for 'SAVE', 'CLEAR CACHE', 'EXPERT', 'LOGOUT', and 'VISUALISATION'. The sidebar menu on the left shows the navigation structure, with 'Call groups' selected. The main content area displays the configuration for 'New call group 1'. The 'Object properties' section includes a 'General information' form with fields for Name, Number, Queue ID, and Timeout. Below this is a 'Members of the call group' table with columns for Name and Number. The 'Connected scenarios' section includes a table with columns for Name, Action, Value, and Order. The 'Outgoing connection' and 'Incoming connection' sections also include tables with columns for ID, Name, Condition, Action, and Value.

Once the above parameters have been configured, it is necessary to add MEMBERS to the call group; in order to do so, please drag the desired objects into the corresponding section of the properties page of the call group. The following object types can be connected to a call group:

- VoIP participants
- External units
- Users (only KNXSERVER)
- Clients (only KNXSERVER)

When connecting a USER, all client devices currently logged with the selected users will be called; if instead you are connecting a CLIENT, the call will go directly to the client device with the corresponding IP address.

Whenever you want to apply changes to the parameters or members of the call group, please hit the button  "UPDATE CONFIG." in order to save them permanently and wait for the spinning circle to disappear:



New call group 1	
New call group 1	
Object properties	
General information	
Name:	New call group 1
Number:	200
Queue ID:	All
Timeout:	30
CLOSE UPDATE CONFIG.	

6 Users (KNXSERVER only)

6.1 INTRODUCTION

The USERS (which grant you access to the VISUALISATION of KNXSERVER) can be used in order to start or receive VoIP calls.

6.2 DIRECT CALLS TO USERS

As already mentioned in chapter 2, all client devices compatible with JAVA will automatically become a VoIP participant as soon as they register with a certain user name through a browser. From this moment, those devices can be reached simply by starting a VoIP call directed to the corresponding user name. If more than one device is online with the same user, the call will be directed to all of them.

Therefore, it is possible to:

- Digit the user name in the intercom pop-up window from another client device (or IP phone / softphone), which supports the input of alphanumeric characters
- Digit the user name into the configuration of a door station / external unit, in order to start the call when one of the pushbuttons of the door station is pressed.
- Digit, in alternative to the user name, the "VoIP NUMBER" of the user to start the call (requires that the VoIP number has been configured in the properties page of the user itself)

The screenshot displays the DIVUS OPTIMA web interface. The top navigation bar includes the logo, version number (2.0.1), and buttons for SAVE, CLEAR CACHE, EXPERT, LOGOUT, and VISUALISATION. The left sidebar shows a menu with options like Setup, Technologies, Customization, Visualisation, Advanced functions, Data-Logging, and Users and permissions. The main content area is titled 'admin' and shows the configuration for a user. The 'Object properties' section includes 'General information' with fields for Username (admin), Password (masked), Repeat password, and Trusted IP (192.168.0.77). The 'VoIP' section has a field for VoIP number (111). Below these are 'CLOSE' and 'UPDATE CONFIG.' buttons. The 'User groups to which the user belongs' section shows a list of groups: Administrators, Powermanagers, and User, each with a plus, minus, and delete icon. At the bottom, there are two empty tables: 'Scenarios commanding the current user' and 'Incoming connection'.

6.3 EVENTS

As already mentioned in the ADMIN MANUAL of KNXSERVER, each user can be connected to EVENTS, in order to execute certain actions on all client devices which are currently logged with the selected user. The possible EVENTS are:

- PAGE JUMP: permits to jump to a certain page within the browser (or APP, if mobile devices) on all devices currently logged with the selected user
- VoIP CALL: permits to start a VoIP call, making all client devices ring which are currently logged with the selected user

The configuration of such an automatic VoIP CALL is very similar to the one seen previously for the VoIP participants (in chapter 3.2.4 of this manual):

- Drag the object that should trigger the EVENT (e.g.: a KNX object) into the section "INCOMING CONNECTIONS"
- Specify the CONDITION that will cause the event to be fired (selecting between the options offered by the object dragged in previously)
- Specify "VoIP CALL" as "ACTION"

In the column "VALUE" it is possible to define the target of the VoIP call, which will be dialed as soon as one of the called client devices takes the call generated from KNXSERVER through the event: it is either possible to input one of the following values directly...

- Number of a VoIP participant
- Number of a call group
- Number of an external unit
- Name of a user or its VoIP number
- Number to be dialed through an external phone line (check out chapter 7)

... or to drag the object that should be called to the corresponding field (after changing the input field into a drop zone using the red button on the right); the following object types are supported:

- VoIP participant
- External unit
- Call group
- User

7 Telephony Lines

7.1 INTRODUCTION

Your KNXCONTROL product permits to manage one or more external phone lines (not part of the LAN network), through which it is possible to start and receive phone calls with / from traditional telephones (both fix and mobile phones). The connection to a phone line can be done in 2 ways:

- Through a VoIP GATEWAY, which converts a traditional line (PSTN / analogue or ISDN) into VoIP
- Through your internet connection, using an external PROVIDER

In both cases, either the GATEWAY or the PROVIDER must be compatible with the SIP protocol.

7.2 CREATION OF A NEW PHONE LINE

In order to create a new PHONE LINE in the OPTIMA interface, please select the entry "TELEPHONY LINES" under "TECHNOLOGIES" → "INTERCOM" and hit the "ADD" button in the TOOLBAR at the bottom. The properties page of the newly created object offers the following parameters for configuration:

NAME	Identifies the phone line within the OPTIMA interface
USER NAME	User name and password to be used as login credentials, both for the registration of a VoIP GATEWAY and for the connection to an external SIP PROVIDER
PASSWORD	
TYPE	Define whether the line should use a GATEWAY or a SIP PROVIDER
ID NAME	Keyword (without spaces or special characters) used by the system to identify the phone line
PREFIX	Specify a number to be used as prefix for external phone numbers (e.g.: "0").
	NOTE: If you are using only ONE phone line, it is not required to specify a prefix: numbers that are not part of the VoIP configuration of OPTIMA are automatically redirected to the external phone line.
PHONE NUMBER	Specify the phone number of the external phone line, including eventual regional codes or additional prefixes
CALL RECIPIENT	Specify the phone number that should start to ring in front of an incoming call from the phone line. This can be: <ul style="list-style-type: none"> The number of a VoIP participant The number of a call group The name of a user or its VoIP number

LEVEL Specify the access level of the line, selecting between "FRIEND" (default), "USER" or "PEER".

If the default setting "FRIEND" should not be enough to complete a correct registration, please refer to the documentation of your GATEWAY / PROVIDER in order to understand the level required to connect to an "ASTERISK" system.

If for the TYPE the entry "SIP" is selected, it is also necessary to configure the following parameters:

HOST IP address or domain name of the SIP PROVIDER

REGISTER Specify whether the connection requires a registration or not (usually this is required if you want to receive calls from the external SIP line)

The screenshot shows the DIVUS OPTIMA web interface. The top navigation bar includes the logo, version (2.0.1), and buttons for SAVE, UPDATE CONFIG, EXPERT, LOGOUT, and VISUALISATION. A search bar is located below the logo. The left sidebar menu is expanded to show 'Telecom line' under 'Telephony lines'. The main content area displays the configuration for a 'Telecom line' with the following fields:

General information	
Name:	Telecom line
User name:	123
Password:	...
Type:	Gateway PSTN/ISDN
ID name:	Gateway
Prefix:	0
Phone number:	04719876543
Call recipient:	200

At the bottom of the configuration area, there are two buttons: 'CLOSE' (red) and 'UPDATE CONFIG.' (green).

Once you configured the necessary parameters, please hit the button "UPDATE CONFIG." in order to save them.

7.3 VOIP GATEWAY CONFIGURATION

If a VoIP GATEWAY is used, it will be necessary to configure it in order to use it in the OPTIMA interface. Generally, the following parameters must be set within the gateway:

- USERNAME and PASSWORD: must be the same as configured in the properties page of the phone line inside your KNXCONTROL device
- IP ADDRESS: IP address of the KNXCONTROL device

Furthermore, the GATEWAYS also contain a series of parameters that handle the redirection of calls and also a set of regional parameters (that must be configured in dependency of the country in which the KNXCONTROL device is used). Further information about this topic usually can be found either in the documentation of the GATEWAY or on specialized internet forums. You can also contact the technical support of DIVUS which can provide some additional hints on this topic.

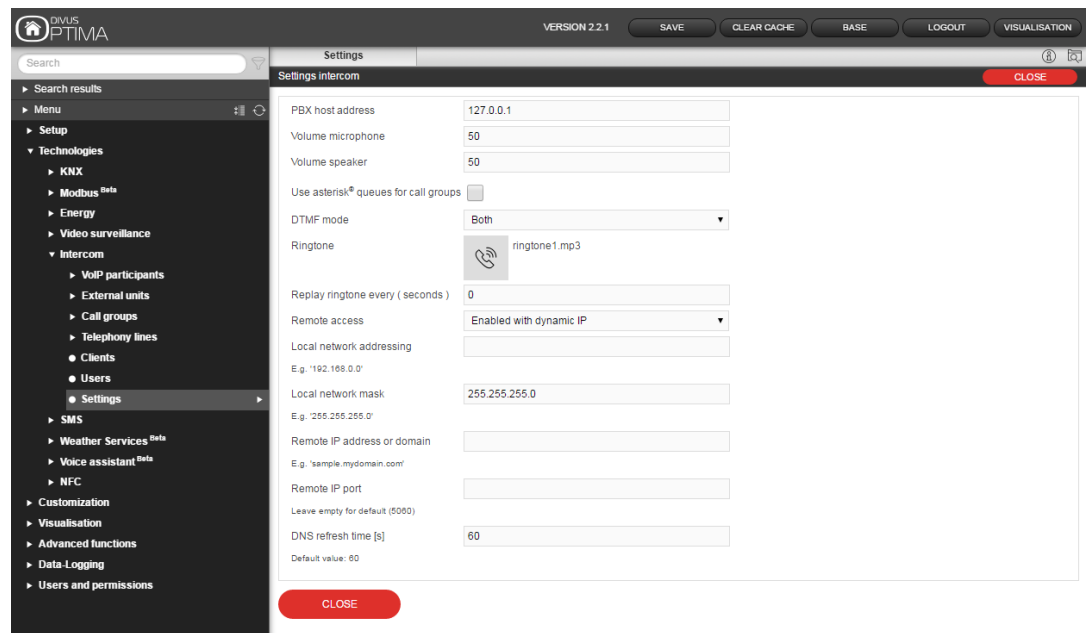
7.4 USE OF THE PHONE LINE

Once the external line has been configured, it is possible to use it to start / receive audio calls to / from traditional phones (fix and mobile phones).

In order to start an OUTGOING call, please dial the desired number – possibly preceded by the “PREFIX”, if defined within the properties page of the phone line – within the intercom pop-up window or from any other VoIP participant.

INCOMING calls from the external phone line will automatically be redirected to the phone number specified in the properties page of the phone line.

8 Settings



8.1 GENERAL

There are these general settings on this page:

PBX HOST ADDRESS	This address defaults to 127.0.0.1 meaning that the device itself should play this role. If you use another server, insert its IP address here. The general intercom settings will then be forwarded to that device.
VOLUME MICROPHONE / SPEAKER (KNX SUPERIO ONLY)	On the KNX Superio you can set the volumes for intercom here
DTMF MODE	Allows to choose how DTMF tones should be sent
RING TONE	Choose the preferred ring tone here
REPLAY RINGTONE	Defines with what frequency the ring tone should be replayed. 0 disables the repeated playing.

8.2 REMOTE ACCESS

If you want to activate the remote access to your intercom system, use these settings:

REMOTE ACCESS	Disabled: disables the remote access function Enabled with static IP: choose this setting if you have a public static IP address Enabled with dynamic IP: choose this setting if you don't have a public static IP address but use a DDNS instead.
LOCAL NETWORK ADDRESSING	Defines the network which should be considered local by the VoIP server. Usually 192.169.0.0 or 192.168.1.0
LOCAL NETWORK MASK	Defines the local network together with the previous setting. Usually 255.255.255.0 or 255.255.0.0
REMOTE IP ADDRESS OR DOMAIN	Depending on the choice for remote access, enter the public static IP address or the domain name which points to your device.
REMOTE IP PORT	Port on your router which forwards data to the sip port of the VoIP server. Preferably don't use 5060 as external port!
DNS REFRESH TIME	Frequency with which the DNS is queried for changes. Used only with DDNS setups. Default is 60.

9 Appendix

9.1 NOTES
