

VIDEOPHONE BOX

User manual

Version 2.1

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

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1 Introduction

1.1 PRESENTATION

The Videophone-Box allows the administration of video intercom functions in combination with the Videophone software. After correct installation and commissioning, the following functions are available:

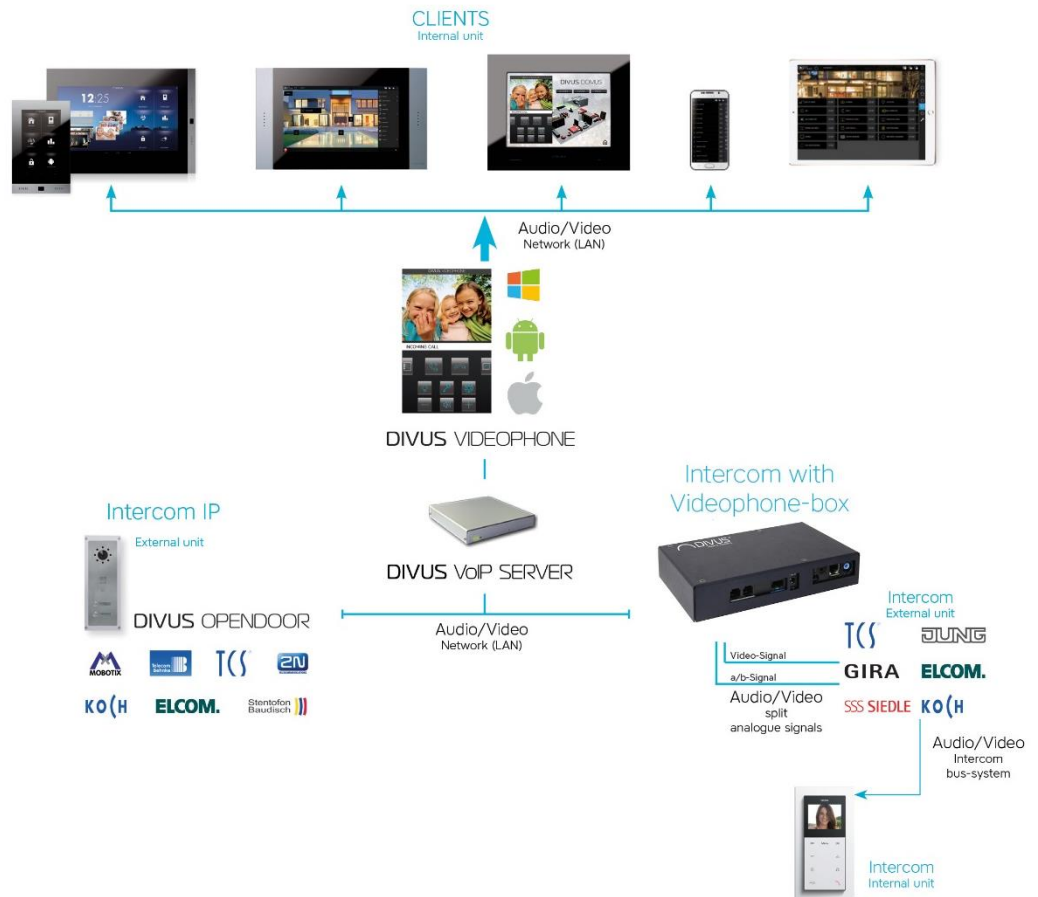
- Receive calls from the external unit of the video intercom on one of the configured client devices and use supported functions like e.g. door opener activation. A client device can be one of the following:
 - One of the DIVUS touch panel models DIVUS Superio, DIVUS Touchzone, DIVUS DOMUS or DIVUS HOME.¹
 - Mobile devices with iOS or Android as operating system with the DIVUS Videophone App.
 - Desktop-PCs or Notebooks
- Communicate between the client devices (Interphone functionality).
- Record voice messages which can be opened by the next user of the device (Messaging-Function)



Note: Detailed information about the Videophone software can be found in the appropriate manual!

¹ The implementation of at least one wall-mounted DIVUS touch panel is recommended for the general reliability of the intercom system.

The following scheme shows what functionalities are offered through the Videophone system:



1.2 IMPORTANT TERMS

For better comprehension of the following instructions and information, some of the most important terms are explained shortly below:

VoIP: "Voice over IP" is an international standard; like the name already lets suspect, it allows speech communication through the network. Initially developed for internet phoning, this technology is used also in other areas, for example for telephone exchange systems or Videophone systems.

VoIP-Server: The VoIP-Server administers the calls of the individual clients present in the network, just like a telephone central; the identification of this VoIP server takes place through the IP address of the device on which the VoIP server is running. In a network, several VoIP servers can coexist without any problems. It is only important that all clients and each software that should communicate among each other, are using the same VoIP server.

Client: as client we define nothing else than a software or a device which is able to register on the VoIP server and afterwards can receive and make calls; the Videophone software for example is such a client. Each client is identified by a 3 digit number (101,102,103...) and can be reached by calling this number.

It is even possible to run the VoIP server and the Videophone software as client on the same PC; the client will run contemporarily with the server without any problems.

Videophone-Box: the Videophone-Box is a hardware, which converts the analog signals of the video intercom to LAN signals and enters into interaction with the VoIP server. Simultaneously, the Videophone-Box makes available the camera picture of the door station as a network stream which can be accessed at any time by the individual clients through the Videophone-Software (provided that the video signal is active permanently).

Videophone software: software for the communication with the intercom system and visualization of the camera picture of the door camera, as well as for the communication of the individual clients among each other.

a/b phone signal: internationally standardized signal (analogue telephone signal), as it is used by customary analogue telephones.

DTMF signal: "Dual-Tone-Multi-Frequency" is a standardized process to transmit the chosen telephone number to the telephone network. Every number / character corresponds to a certain tone which consists of several frequencies and is recognized from standardized switchboards and the telephone network. The a/b transformer modules, to which the Videophone-Box is connected, also use this technology in order to execute an action over the conveyed DTMF signal (for example to activate the door opener). Usually only short sign combinations, mostly even just individual numbers, are used on that occasion. The a/b transformer module of BPT for example opens the door if it receives the number "2" from the client as a DTMF signal during an active communication.

Network stream: essentially it is a data flow that can be accessed through the network. In case of the Videophone-Box we are speaking about still image collected from the Videophone-Box that are called from the Videophone software as data flow and are displayed in the main window of the Videophone software as a video.

Backend: main process of the Videophone software. It comprises the entire functionality of the software and runs in the background. At incoming calls it can be noticed, as the video windows fades in and the ringtone is activated.

GUI: „Graphical User Interface“. Term for a graphical interface through which the user can command the software. Also the Videophone software has an integrated "GUI".

Frontend: the frontend of the Videophone software is the integrated "GUI". Over this graphical interface the user can configure the software and use its various functions.

UDP: „User Datagram Protocol“. UDP is a simplified network protocol which can be used to remotely control the Videophone software. For example, it enables you to create your individual "GUI" which can be used instead of the frontend of the Videophone software.

1.3 SYSTEM REQUIREMENTS

In order to be able to use the whole functionality of a Videophone system, the following devices must be present in the installation:

- At least one DIVUS touchpanel (DIVUS Superio, DIVUS Touchzone, DIVUS DOMUS or DIVUS HOME with multimedia.) These devices already contain the activated Videophone software.
- VoIP-Server (can also run on one of the installed PCs as software).
- A Videophone-Box
- Intercom outdoor station - might need additional modules to make it compatible with the Videophone-Box input signals.

The Videophone-Box offers compatibility to all video intercom systems on the market which meet the following requirements:

- The intercom system must be able to provide the following signals in front of an incoming telephone call from an external unit of the system (either directly over an analogue BUS signal or over a corresponding converter interface):
- Analogue a/b phone signal (international analogue telephone standard)
- Analogue video signal (RCA 75Ohm, 1Vpp, like mostly used on camcorders)

It must be possible to transfer a telephone call from the external unit of the video intercom system using the above named signals, so that the call - after the signal conversion over the Videophone-Box - can be handled from the VoIP server. After acceptance of the telephone call (through a DTMF signal), both the video signal and the bidirectional audio signal must be available without any further action.

The video intercom systems existing on the market principally differ in the transfer type of the audio and video data: there are analogous and digital systems. In the case of digital systems, which can be found more and more, there usually exist corresponding converter interfaces, which make available the implied signals. In the appendix of this manual you can find a list of some compatible intercom systems that already have been tested by DIVUS in combination with Videophone. For more information about compatibility of certain intercom systems with Videophone or an updated list of the tested products, please contact the technical support of DIVUS.

1.4 CONTENT OF THE PACKAGE

The Videophone-Box is delivered with the following components:

- Videophone-Box
- Accessories:
 - Power supply Audio (VoIP)
 - Power supply Video
 - Network cable RJ45 Audio (VoIP)
 - Network cable RJ45 Video (Videosever)
 - 2 wire phone cable RJ11 for connecting the Videophone-Box with the a/b phone signal of the intercom system
 - RCA cable / plug for connecting the Videophone-Box with the video signal of the intercom system
 - Product sheet containing the default settings of the Videophone-Box (for example the IP addresses, user credentials etc.)

2 Wiring and configuration

2.1 INTRODUCTION

For a better comprehension, please read the following short explanation on how the Videophone-Box works:

The Videophone-Box is only used to convert the analogue signals of the intercom system to network-capable signals, so that the VoIP server can handle the signals. In order that the individual clients can communicate among each other and with the video intercom, all must register on the same VoIP server.

The analogue a/b telephone signal is converted into an international VoIP-Standard, through which the VoIP server can create communication with the intercom system. The analogue video signal is converted into a network video stream by the Videophone-Box, which can be shown at any time from the Videophone software. The software combines the audio and video signals, what makes possible visual communication with the person who is ringing at the door.

As soon as the button on the external unit is pressed, a corresponding dial module or a/b transformer module which is connected to the Videophone-Box over a telephone cable, dials out a number. The Videophone-Box recognizes this number and makes a connection to the PC with the active VoIP server over the network. The VoIP server executes the actions which were defined for the number dialled by the a/b transformer module; normally, a connection to several clients is established. As soon as the connection to the clients is up, the Videophone software appears automatically on the client PCs, showing the video signal of the external unit and emitting the ringtone over the built-in loudspeaker.

So there is an acoustic signalization whenever anybody rings at the door and additionally the video window shows who the person in front of the door is and what it is doing.

By clicking on the "Accept" button, the call will be activated and the audio communication between the client and the external unit of the intercom system will be established. For establishing this communication so-called DTMF tones are transmitted to the intercom system. In the same way, but with different DTMF tone sequences, the door opener or the door light can be controlled. Also, the connection can be terminated by the Videophone software by clicking on the related button of the frontend.

In few words the Videophone-Box can be seen as interface between client PCs with Videophone software (network) and intercom systems (analogue signals).

Furthermore, the Videophone software enables calls among the single clients (Interphone function).

2.2 WIRING

As already mentioned, the Videophone-Box is divided in 2 parts:

Audio: A VoIP gateway converts the analogue a/b signal from the intercom system into standard VoIP packages. The required connections are the following:

1. Connect the power supply Audio to a 230V power socket
2. Connect the RJ11 phone cable with the a/b phone line of the intercom system
3. Connect the RJ45 network cable Audio (VoIP) with the local network (Switch/Router)

Video: A video server converts the analogue video signal from the intercom system into a network stream. The required connections are the following:

4. Connect the power supply Video to a 230V power socket.
5. Connect the RCA video cable to the analogue video signal of the intercom system.
6. Connect the RJ45 network cable to the local network (Switch/Router)



2.3 CONFIGURATION OF THE INTERCOM SYSTEM

The video intercom system must be configured in a way that – in presence of a telephone call from the external unit – the a/b phone signal and the video signal are activated by the conversion modules of the intercom system. Normally such conversion modules are handled like internal units inside the intercom bus system.

2.4 AUDIO CONFIGURATION (VOIP)

2.4.1 INTRODUCTION

The VoIP gateway of the Videophone-Box converts the analogue a/b phone signal into VoIP signals, which can be evaluated by the VoIP server and afterwards redirected to the different client PCs. As shown below, the gateway has 4 communication ports:

PORT	ANSCHLUSSTYP	FARBE	FUNKTION
LINE 1	RJ11	Dark green	Not used
LINE 2	RJ11	Light green	Connection to the a/b phone signal from the intercom system (use only the inner two contacts of the RJ11 plug)
ETHERNET	RJ45	Yellow	Not used
INTERNET	RJ45	Blue	Connection to the network, for communication with the VoIP server and the Videophone software and at the same time for remote configuration purpose

The configuration menu of the gateway can be accessed over the blue INTERNET port from every PC whose IP address is compatible with the IP address of the gateway itself.



2.4.2 PREPARE THE NETWORK CONNECTION

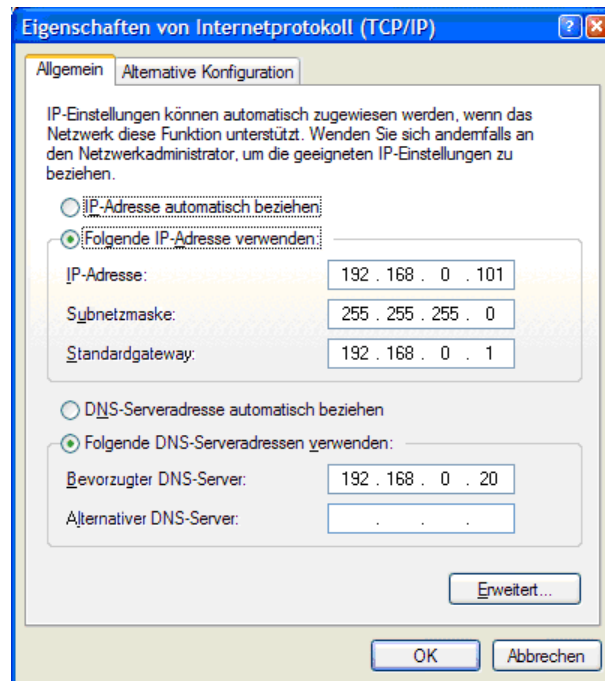
In order to access the gateway configuration menu:

1. Ensure that the blue INTERNET port of the VoIP gateway and the PC, over which you want to configure the gateway, are connected to the local net-work. The PC and the VoIP gateway can also be connected directly; in this case a crossover network cable must be used to connect the devices. Furthermore check that the power supply of the VoIP gateway is plugged in.



Important: don't connect the VoIP gateway to your network before you configured it in accordance to your network.

2. Change the IP address of the PC you are using to a value, which is compatible with the IP address of the VoIP gateway; the illustration beneath shows an example for a VoIP gateway with IP address 192.168.0.120. It's important that the first three blocks of the IP address of the PC are identical to the first three blocks of the IP address of the VoIP gateway (in this case 192.168.0). The fourth block completes the IP address and must be one of the free values in the network (for example 101). For changing the IP configuration of the PC, the network configuration window of the used network connection must be opened; there the entry "Internet Protocol (TCP/IP)" (or similar) can be selected and opened by clicking on "Properties".

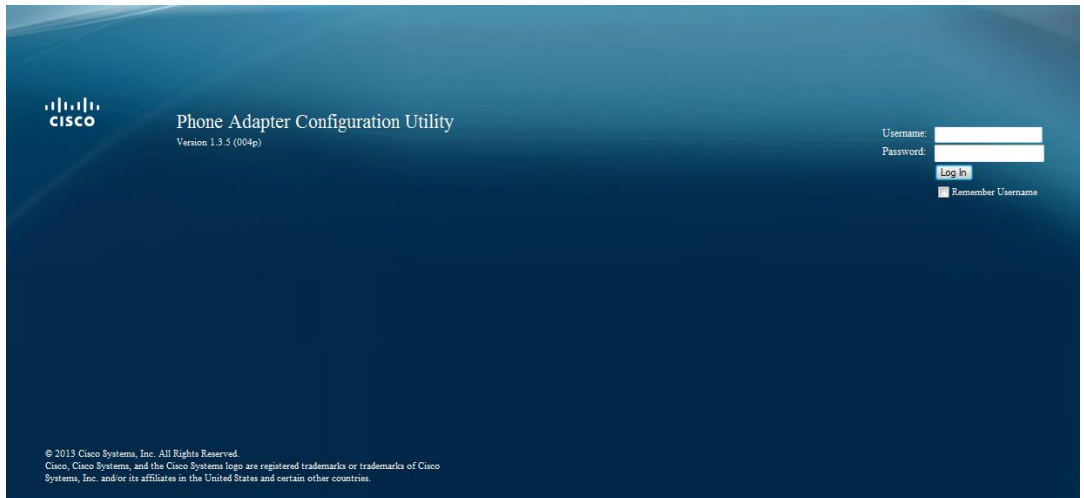


2.4.3 CONFIGURATION WITH WEB-INTERFACE

Open your internet browser and type the following URL into the address bar:

`http://<IP address of the VoIP gateway>`

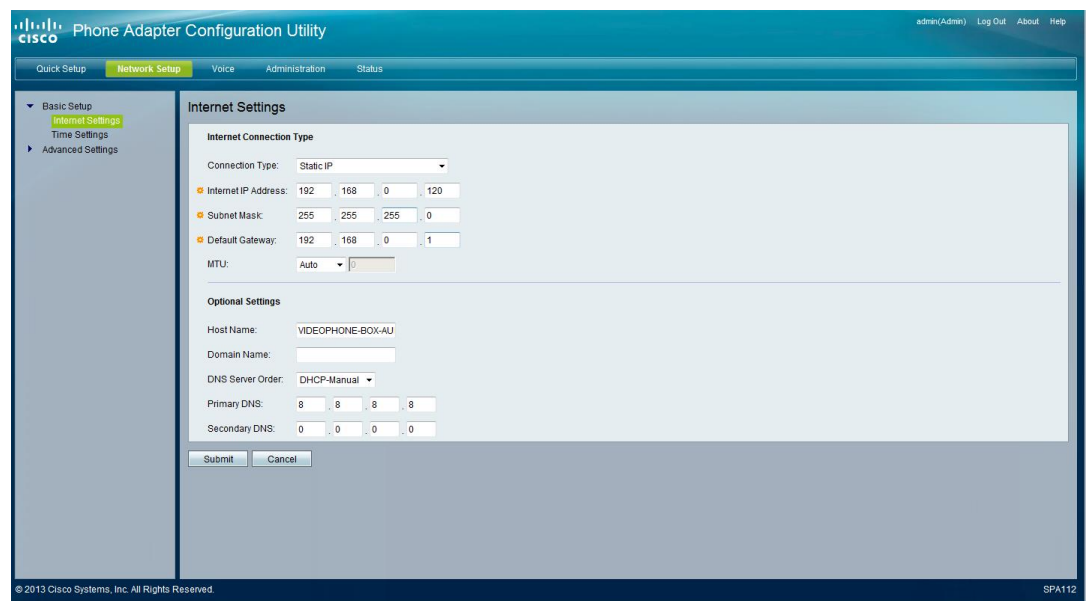
For default configuration: <http://192.168.0.120/>



You will see the login page of the VoIP-Gateway.

The default username is "admin" and the default password is "1234".

Once logged in, you will see the "quick setup" menu, navigate to the "Network Setup" menu.



Use "Static IP" as connection Type. The default IP-address is 192.168.0.120, but you can change the IP address of the VoIP-gateway itself by changing the "IP Internet Address"; don't forget to check if the "Subnet Mask" is compatible with the new IP address.

To save the changes made in this menu click on "Submit"; the VoIP gateway will reboot and will then be reachable on the new IP address.

Now go back to the "Quick Setup" page.



Note: The default line is Line 2.

The screenshot shows the Cisco Phone Adapter Configuration Utility (SPA112) interface. The main content area is titled "Quick Setup" and contains two sections for Line 1 and Line 2. Line 1 fields are empty. Line 2 fields are pre-filled: Proxy (192.168.0.100), Display Name (901), Password (masked with asterisks), User ID (901), and Dial Plan (<S0:dooropener0-xxx@192.168.0.100>). The bottom of the form has "Submit", "Cancel", and "Refresh" buttons. The footer includes "© 2013 Cisco Systems, Inc. All Rights Reserved." and "SPA112".

"Proxy": please insert the IP address of the PC on which the active VoIP server is running or the IP address of the VoIP server (for example 192.168.0.100).

"Display Name", **"User ID"**, and **"Password"**: In this fields the authentication data, which the gateway should use to register on the VoIP server, must be specified. These parameters must all use the same value; you can freely choose between the pre-defined values 901, 902, 903, 904, 905, 906, 907, 908, 909, 910. Nevertheless, it is recommended to start with 901. Those values stand for the 10 pre-configured external intercom units supported by Videophone.

"Dial Plan": this is the string through which the VoIP server is informed about which destination device to call when a button is pushed. The first thing to know is whether the outdoor station in use is capable of sending no, one or more 3 digit numbers when pushing its button(s).

1. It can send one number only (or none). In this case, the number to call can be inserted directly into the "dial plan" string. As an example we'll take number 111.
 - a. For DIVUS VS1 with IP address 192.168.0.123 the string to insert is <S0:dooropener0111@192.168.0.123>
 - b. For DIVUS KNX Server with IP address 192.168.0.234 the string to insert is <S0:111@192.168.0.234>
2. It can send 2 or more 3 digit numbers (like e.g. Siedle). In this case the outdoor station has to be configured to call the 2 (or more) chosen numbers using its own setup software or web interface.

- a. For DIVUS VS1 with IP address 192.168.0.123 the string to insert is
<S0:dooropener0>xxx<:@192.168.0.234>
- b. For DIVUS KNX Server with IP address 192.168.0.234 the string to insert is
<S0:>xxx<:@192.168.0.234>

To save all changed parameters, click on „Submit“.



Note: The VS1 has a set of predefined extensions (call functions) which use the *dooropener* context. It is recommended to use those extensions and – as a consequence – the *dial plan* strings as shown above. If you need to use custom call numbers, you may add them by editing the *extensions_custom.conf* file using the web interface; the easiest way is by copying/pasting an existing extension and then just changing the call number. You then have to change the “dial plan” string according to your new entry.



Note: VS2 and VS3 use the same settings as VS1. Other VoIP servers may use the same settings as the DIVUS KNX Server (possibly without the brackets <...>).

If you are unable to reach other clients after the completion of the “Quick Setup”, please go to “Voice” and select “Line 2” in the submenu

Information System SIP Provisioning Regional Line 1 User 1 **Line 2** User 2

Line 2

General
 Line Enable: YES

Streaming Audio Server (SAS)
 SAS Enable: no SAS DLO Refresh Intvl: 30
 SAS Inbound RTP Sink:

NAT Settings
 NAT Mapping Enable: no NAT Keep Alive Enable: no
 NAT Keep Alive Mig: \$NOTIFY NAT Keep Alive Dest: \$PROXY

Network Settings
 SIP ToS/DiffServ Value: 0xb8 SIP CoS Value: 3 [0-7]
 RTP ToS/DiffServ Value: 0xb8 RTP CoS Value: 6 [0-7]
 Network Jitter Level: High Jitter Buffer Adjustment: yes

SIP Settings
 SIP Transport: UDP SIP Port: 5060
 SIP 100REL Enable: no EXT SIP Port:
 Auth Resyno-Reboot: yes SIP Proxy/Require:
 SIP Remote-Party-ID: yes SIP GUID:
 SIP Debug Option: 1-line RTP Log Intrl: 0
 Restrict Source IP: no Referor Bye Delay: 4
 Refer Target Bye Delay: 0 Referee Bye Delay: 0
 Refer-To Target Contact: no Sticky 183: no
 Auth INVITE: no Reply 182 On Call Waiting: no
 Use Anonymous With RPID: yes Use Local Addr In FROM: no

Call Feature Settings
 Blind Adv-User Enable: no MCH Server:
 Xfer When Hangup Conf: yes Conference Bridge URL:
 Conference Bridge Ports: 3 Enable IP Dialing: no
 Emergency Number: Mailbox ID:
 Feature Key Sync: no

Proxy and Registration
 Proxy: 192.168.0.100
 Outbound Proxy:
 Use Outbound Proxy: no
 Register: yes Use OB Proxy In Dialog: no
 Register Expires: 3600 Make Call Without Reg: no
 Use DNS SRV: no Ans Call Without Reg: no
 Proxy Fallback Intrl: 3600 DNS SRV Auto Prefx:
 Mailbox Subscribe URL: Proxy Redundancy Method: Normal
 Mailbox Subscribe Expires: 2147483647

Subscriber Information
 Display Name: 801 User ID: 801
 Password: ***** Use Auth ID: no
 Auth ID: Resident Online Number:
 SIP URI:

Supplementary Service Subscription

Call Waiting Serv:	<input checked="" type="checkbox"/> yes	Block CID Serv:	<input checked="" type="checkbox"/> yes
Block ANC Serv:	<input checked="" type="checkbox"/> yes	Dist Ring Serv:	<input checked="" type="checkbox"/> yes
Cfwd All Serv:	<input checked="" type="checkbox"/> yes	Cfwd Busy Serv:	<input checked="" type="checkbox"/> yes
Cfwd No Ans Serv:	<input checked="" type="checkbox"/> yes	Cfwd Sel Serv:	<input checked="" type="checkbox"/> yes
Cfwd Last Serv:	<input checked="" type="checkbox"/> yes	Block Last Serv:	<input checked="" type="checkbox"/> yes
Accept Last Serv:	<input checked="" type="checkbox"/> yes	DND Serv:	<input checked="" type="checkbox"/> yes
CID Serv:	<input checked="" type="checkbox"/> yes	CW/CID Serv:	<input checked="" type="checkbox"/> yes
Call Return Serv:	<input checked="" type="checkbox"/> yes	Call Redial Serv:	<input checked="" type="checkbox"/> yes
Call Back Serv:	<input checked="" type="checkbox"/> yes	Three Way Call Serv:	<input checked="" type="checkbox"/> yes
Three Way Conf Serv:	<input checked="" type="checkbox"/> yes	Attn Transfer Serv:	<input checked="" type="checkbox"/> yes
Unattn Transfer Serv:	<input checked="" type="checkbox"/> yes	MWI Serv:	<input checked="" type="checkbox"/> yes
VMI Serv:	<input checked="" type="checkbox"/> yes	Speed Dial Serv:	<input checked="" type="checkbox"/> yes
Secure Call Serv:	<input checked="" type="checkbox"/> yes	Referral Serv:	<input checked="" type="checkbox"/> yes
Feature Dial Serv:	<input checked="" type="checkbox"/> yes	Service Announcement Serv:	<input type="checkbox"/> no
Reuse CID Number As Name:	<input checked="" type="checkbox"/> yes	CONFID Serv:	<input checked="" type="checkbox"/> yes

Audio Configuration
 Preferred Codec: G711u Second Preferred Codec: G711a
 Third Preferred Codec: Unspecified Use Pref Codec Only: no
 Use Remote Pref Codec: no Codec Negotiation: Default
 G729a Enable: yes Silence Supp Enable: no
 G729-32 Enable: yes Silence Threshold: medium
 FAX V21 Detect Enable: yes Echo Canc Enable: yes
 FAX ONG Detect Enable: yes FAX Passthru Codec: G711u
 FAX Codec Symmetric: yes DTMF Process INFO: yes
 FAX Passthru Method: NSE DTMF Process AVT: yes
 FAX Process NSE: yes DTMF Tx Method: InBand

Audio Configuration
 Preferred Codec: G711u Second Preferred Codec: G711a
 Third Preferred Codec: Unspecified Use Pref Codec Only: no
 Use Remote Pref Codec: no Codec Negotiation: Default
 G729a Enable: yes Silence Supp Enable: no
 G729-32 Enable: yes Silence Threshold: medium
 FAX V21 Detect Enable: yes Echo Canc Enable: yes
 FAX ONG Detect Enable: yes FAX Passthru Codec: G711u
 FAX Codec Symmetric: yes DTMF Process INFO: yes
 FAX Passthru Method: NSE DTMF Process AVT: yes
 FAX Process NSE: yes DTMF Tx Method: InBand
 FAX Disable ECAN: no DTMF Tx Mode: Strict
 DTMF Tx Hold Off Time: 70 FAX Enable T38: yes
 Hook Flash Tx Method: None FAX T38 Redundancy: 1
 FAX T38 ECM Enable: yes FAX Tone Detect Mode: caller or callee
 Symmetric RTP: yes FAX T38 Return to Voice: no
 Modem Line: no RTP to Proxy in Remote Hold: no

Dial Plan
 Dial Plan: <S0:doorphone0>xxxx@192.168.0.100>

FXS Port Polarity Configuration
 Idle Polarity: Forward Caller Conn Polarity: Forward
 Callee Conn Polarity: Forward

Submit Cancel Refresh

- "Line Enable": please set this value on "yes" to activate the VoIP functionality of the gateway
- "SIP Port": please choose the port, over which the gateway should communicate with the VoIP server. The default value is "5060".
- "Proxy" and "Outbound Proxy": please insert the IP address of the PC on which the active VoIP server is running or the IP-address of the VoIP server (for example 192.168.0.100).
- Fill the fields "Display Name", "User ID", "Password" and "Dial Plan" as described above.

Save the changes by pressing "Submit".

The submenu "Information" should now display "Registered". That means that the VoIP-Gateway is registered to the VoIP-server. If not, please check your settings.

The screenshot shows the Cisco Phone Adapter Configuration Utility web interface. The top navigation bar includes 'Quick Setup', 'Network Setup', 'Voice', 'Administration', and 'Status'. The left sidebar contains a menu with 'Information' selected, and sub-items for 'System', 'SIP', 'Provisioning', 'Regional', 'Line 1', 'User 1', 'Line 2', and 'User 2'. The main content area is titled 'Information' and displays configuration details for Line 2. The 'Registration State' is highlighted in red and shows 'Registered'. Other visible details include 'Hook State: Off', 'Last Registration At: 7/12/2016 23:33:26', 'Call 1 State: Proceeding', and 'Call 1 Type: Outbound'.

If you configured the VoIP gateway directly over your PC (by connecting it with a crossover network cable), you can now disconnect the VoIP gateway from the PC and connect the blue INTERNET port of the VoIP gateway to the local network with a normal network cable (RJ45). Now the VoIP gateway is ready for use; you can test if the VoIP gateway is configured correctly by following the steps below:

- Open the start menu of a Windows PC connected to the local network and choose „Run“
- Enter "cmd" and [ENTER] in the appearing window to open the „Command prompt“
- Type "ping 192.168.0.120" (IP address of the VoIP gateway) and hit „Enter“.
- Wait a moment and check if the VoIP gateway answers to the "ping"-command. If the VoIP gateway does not answer, there is something wrong; either the VoIP gateway is not configured correctly, or it is not connected properly.

2.5 VIDEO CONFIGURATION

2.5.1 CONNECTIONS

- Connect the video signal of the installed door camera to the first input of the video server.
- Plug in the power supply and connect it to the power jack of the video server; ensure that the power switch beneath the power jack is turned on.



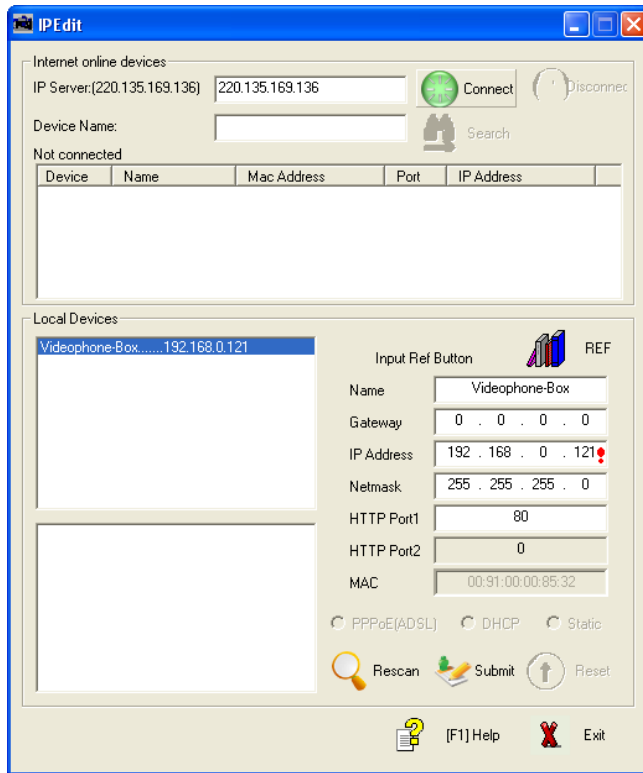
2.5.2 VIDEO SERVER NETWORK SETTINGS

Please insert the Videophone CD into your PC and start the program IPEDIT, which can be found in the folder „Tools“.

If the video server is already connected to the network, its IP address should be listed under "Local Devices". If not, connect the video server to the local network and click on "Rescan" for rescanning the network.

By double-clicking the IP address of the video server you will be redirected to its configuration page, where you can change various settings described in the consequence. In alternative, you can even open a browser window: by inserting the IP address of the video server you will be redirected to its configuration page, too.

The default IP address of the video server can be found on the product insert (normally 192.168.0.121)

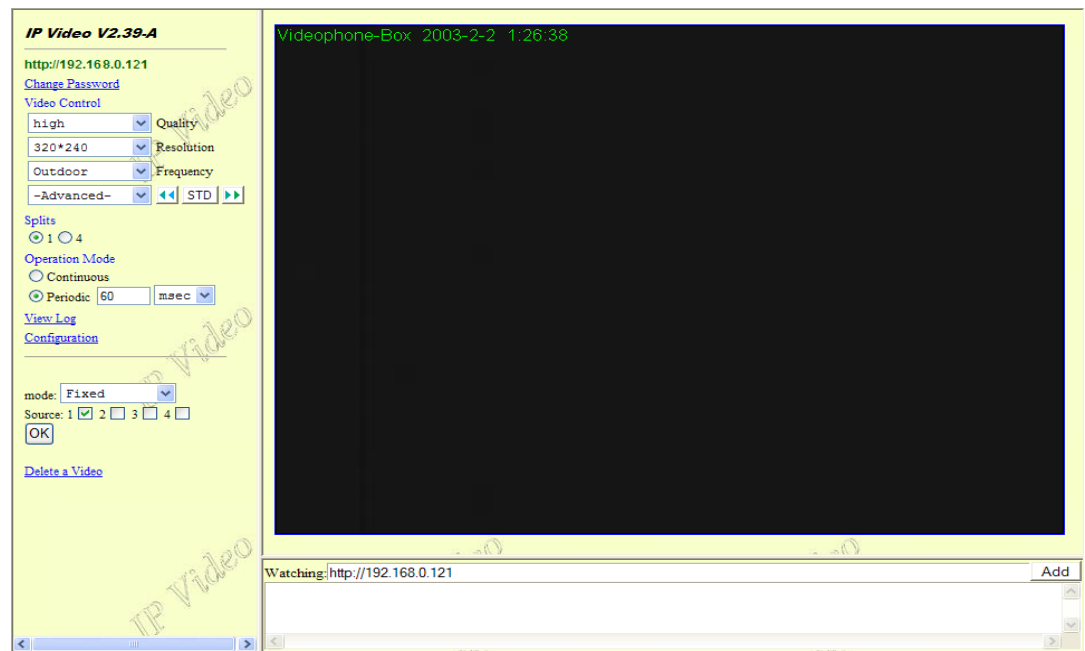


Hint: Please ensure that the network cable (LAN, RJ45) is connected to with the video server. Check out that the video server is switched on (power switch beneath the power jack). If the video server works correctly on the local network, the green LED on the network port of the video server will start blinking or lighten up.

The video server can be configured with Internet Explorer as well as with Firefox or other similar browsers. However, it's recommended to use Internet Explorer, because it is compatible with all functionalities of the video server.

2.5.3 CONFIGURATION WITH WEB-INTERFACE

As soon as the configuration menu of the video server is opened within a browser or with the IPEDIT tool, the following screen will appear:



Hint: The graphical interface of the configuration page itself depends on the used browser. The screenshots that are shown in this manual refer to Microsoft Internet Explorer, which guarantees the full access to all settings / functions of the video server.

In the left column, the primary options of the video server can be modified. The available parameters are:

- **Quality:** compression level of the video signal; a higher quality also causes a larger quantity of data, which has to be transferred and consequently elaborated on the network.
- **Resolution:** resolution of the camera picture, adjustable in pixel (width x height); please choose one of the supported values.
- **Frequency:** frequency of the incoming video signal, depends on the signal of the attached cameras (PAL / NTSC / others).
- **Splits:** defines if only one video signal (full screen) or 4 video signals (split screen) should be shown.
- **Operation mode:** allows to set up a refresh rate; in dependency of this value the pictures of the cameras are updated.
- **Mode:** this setting defines whether the shown video picture should be fixed (there will only be shown the image of the defined channel) or whether should be changed between the different channels (rotation mode, "round robin"); in the first case it can be defined which one of the 4 channels should be shown.



Important! Use only the „fixed“ parameter for the „mode“ settings, otherwise in Videophone there will be no continuous video stream.

In the central part of the configuration window, the current video signal is shown in dependency of the above listed settings. The lower part of the screen records messages of the video server, as for example movement notifications in case of active motion detection function. In this window can be supervised even more than one video server: by clicking the button "Add", the IP addresses of other video servers can be inserted.

By clicking on the link "Configuration" in the left column of the screen, further settings of the video server can be edited:

System Settings

Over this page the system settings of the video server, like for example the device name (which is used to identify the video server within the network), the system time and time zone as well as the ports of the HTTP connection and the video stream, can be modified. The default port for the HTTP connection is "80"; this setting should only be changed if absolutely necessary (for example if the signal of the attached cameras as well as the configuration page itself should be visible through internet by enabling the port forwarding function in the DSL router).

User Settings

In this configuration window can be defined whether an authentication is necessary to view the camera images or not. If an authentication should be required, a username and a password can be inserted; if necessary, it is also possible to delete already existing users / passwords.

Motion Detect

This page contains the administration menu of the so called "motion detection" (automatic recognition of movements). To use this function, it must be activated in this page and there must be set the recognition rate (Attention, a higher sensitivity of the recognition rate requires a higher refresh rate of the camera image and could increase the sensitivity of the signal in presence of disturbances).

If a movement is recognized, the following actions can be executed by the video server:

- The forwarding of a mail with the recorded camera pictures as attachment
- The transfer of the recorded camera pictures to a predefined FTP server

In the section "Mail settings" the parameters for the configuration of the mail box can be set, which are necessary to be able to send out mails. Among others, the server (SMTP) and the log-in data of the used mail box as well as the desired recipient's addresses must be specified. In order to send out a mail at the recognition of a movement, the option "send mail when motion detected" must be activated.

The section "FTP settings" however serves to the configuration of the transfer of recorded camera pictures onto a FTP server. In this case, the FTP Server itself, the log-in data and the account name (dependent on the configuration of the FTP server) have to be configured. The option "upload path" permits to insert the target folder on the FTP server, in which the camera pictures should be stored. By changing the value "interval" the time distances, in which the pictures should be copied, can be modified. As already seen in the mail settings, even here an option called "upload images when motion detected" has to be activated in order to be able to use the just described function.

Network Settings

Over this configuration page the network settings of the video server can be changed. Primarily can be set whether the video server should get its IP address over DHCP or whether a static IP address should be used (default setting). In the second case the necessary data must be inserted manually.

Furthermore, the DNS parameters for the internet access can be defined, which are necessary for the correct transfer of the camera pictures via mail / internet.

Over the section "IPMail" the video server can be configured in a way that it sends out periodically an e-mail with its own IP address. This function can be very useful if the IP address of the video server changes dynamically (what in case of a DHCP network can absolutely be, if not regulated separately).

DDNS Settings

The page "DDNS" can be used in order to handle the video server over the free DNS service "DynDNS" (www.dyndns.org). This service - after free registration at www.dyndns.org - allows a constant remote access to the video server over internet even with changing IP address. If this service is activated, the video server sends its IP address to the DynDNS service in regular intervals; if the service recognizes an alteration of the address, it handles the redirection of the signals automatically, so that the camera signal is never interrupted. Further information about this function can be found on the home page of DynDNS.

2.6 2.6 FINAL CHECK OF THE CONNECTIONS

Before the different components will be used in combination with the Videophone software, it is recommended to make the following tests, in order to guarantee that all wirings, connections and configurations are correct and working:

Video:

- Please open a new browser window and insert the IP address of the video server
- Now start a call on the external unit of the intercom system
- Check now if the video signal of the door camera is shown correctly in the Browser window. Configure the correct input channel or use the option "quad-view" to view all four channels at the same time. With some intercom systems the video signal will only be activated when a call is launched!

Audio:

- Please open a new browser window and insert the IP address of the VoIP gateway
- Jump to the menu „Voice“
- Open the menu „Information“ and refresh the page by hitting „F5“ or selecting "Refresh" in the browser menu.

- Please check if in the section „Line 2 Status“ the „Registration State“ is set to „Registered“. If there is no value or a different value (like e.g. „Registration failed“), the VoIP gateway did not complete the registration on the VoIP server. In this case, please check the connections and settings of the audio part again!



Hint: A custom analogue telephone can be of big benefit for troubleshooting. If it is connected to the Videophone-Box instead of the a/b transformer module, the functionality of the Videophone system can be tested with the telephone. Simply lift the receiver, and dial the required number (for standard configuration 101).

As a result, the active clients should begin to ring and Videophone software should appear. This method can be used as a very quick diagnosis which identifies whether a possible malfunction should be on Videophone or on intercom side.



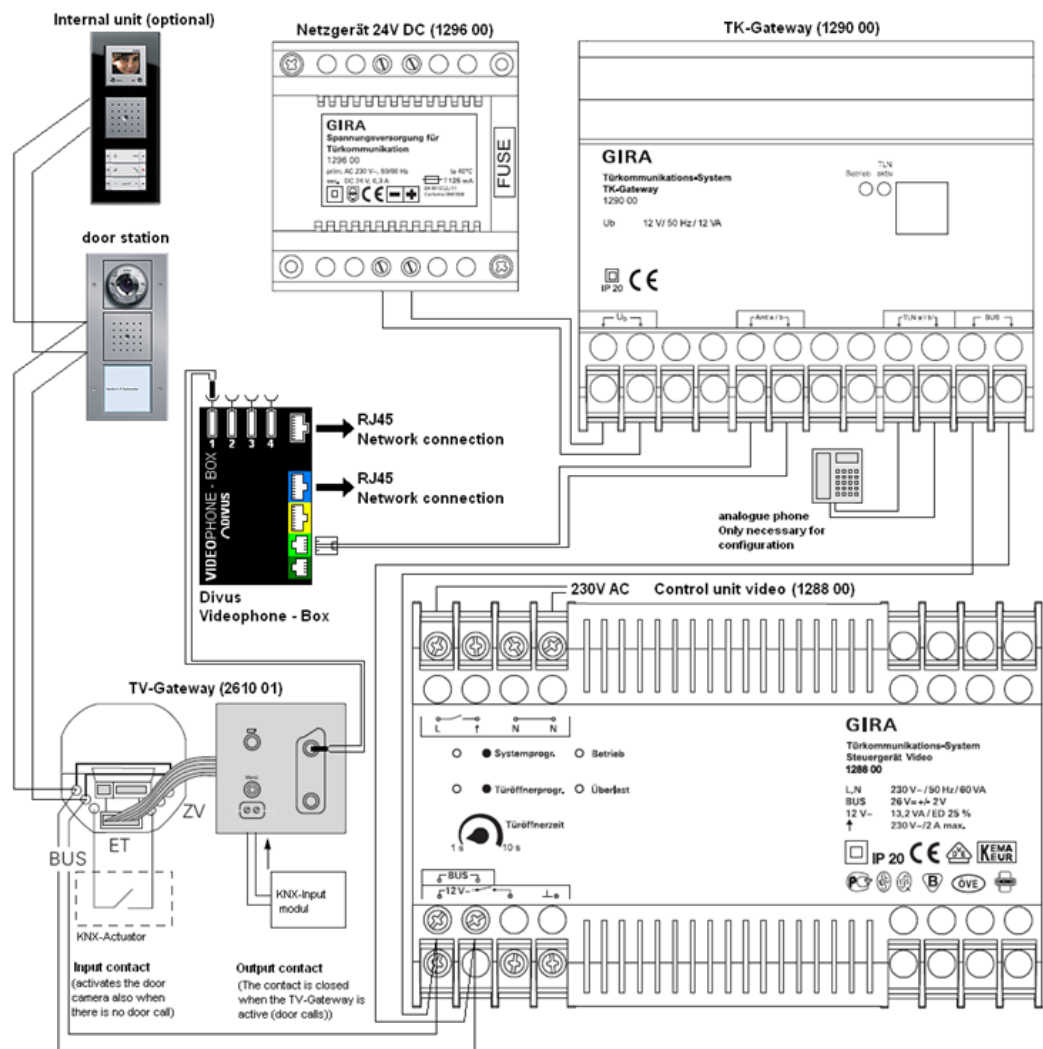
Important: The tests mentioned on this side refer to the entire Videophone system! It is necessary that - beside the Videophone-Box - a VoIP server from DIVUS is present in the network!

If the tests listed above cannot be concluded in a positive way, please check the settings of the devices as well as their connections again by confronting them to the values shown within the last chapters. However, if the problems still remain, please refer to the technical support of DIVUS.

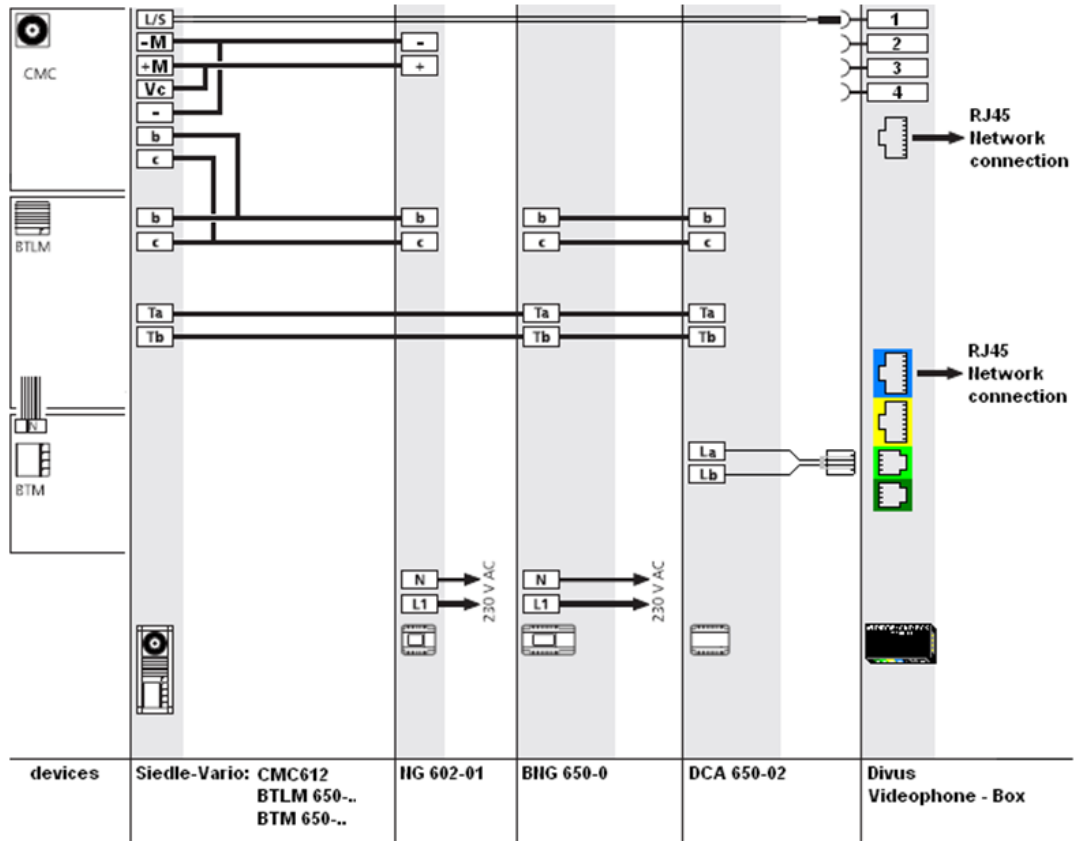
3 Appendix

3.1 CONNECTION SCHEMES OF COMPATIBLE INTERCOM SYSTEMS

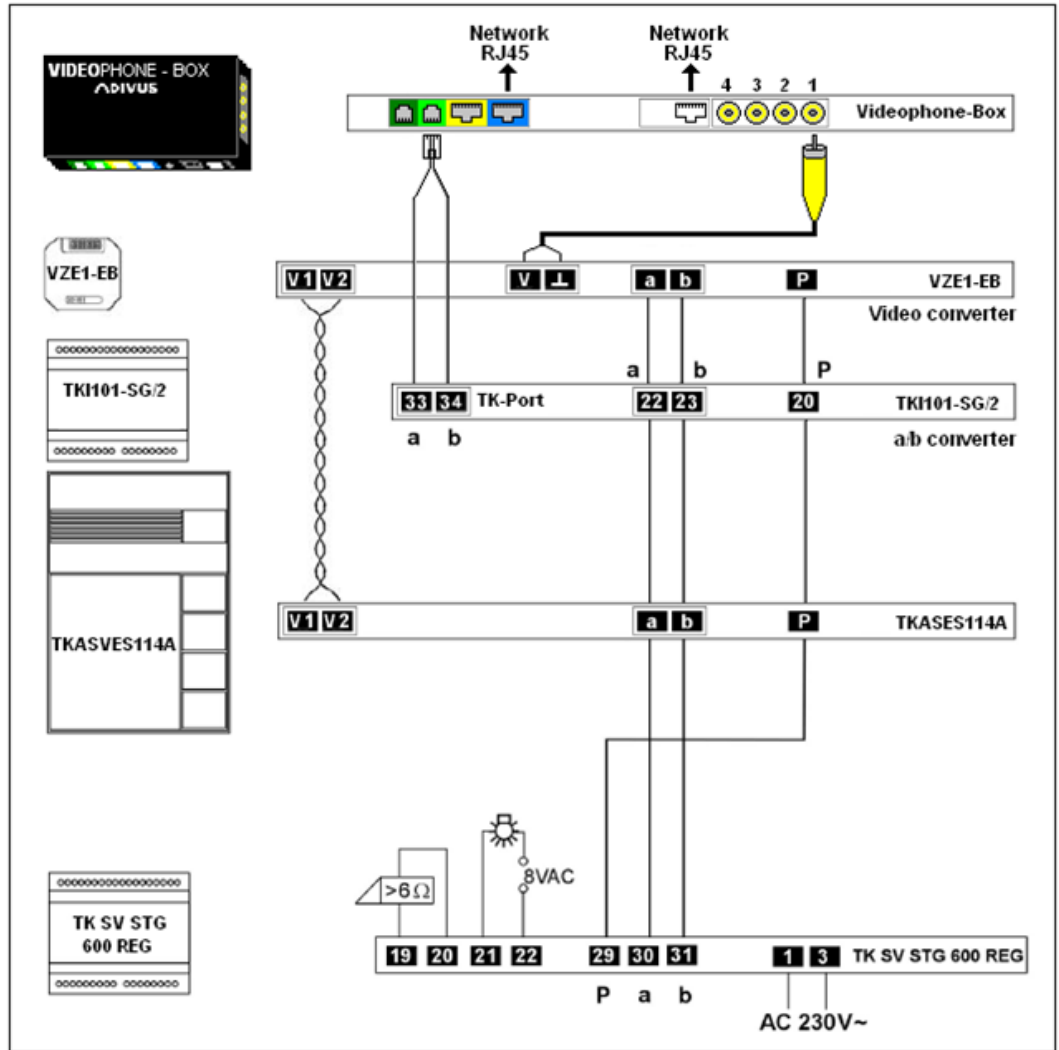
GIRA



SIEDLE



JUNG



TCS

