LANCOM VoIP Router

- Handbuch TK-Anlagenfunktionen
- Manual PBX Functionalities

LANCOM Systems

LANCOM VoIP Routers
PBX Functionalities

LANCOM
Systems

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Preface

Thank you for your confidence in us!

LANCOM Business-VoIP-Routers offer innovative all-round solutions that integrate data and voice applications with basic PBX functions in a single compact device. The range of interfaces available enables a LANCOM Business-VoIP-Router to provide connections for ISDN or analog telephones, and fax machines. The LAN interfaces allow additional SIP telephones or softphones in the internal network to be connected up as well.

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LANCOM Routers provide PBX functions via Voice over IP, making them a technically mature alternative to conventional ISDN PBXs. They are ideal for smaller companies, branch offices and telecommuters. Even if the Internet connection should fail, calls can still be made to the PSTN via an ISDN or analog interface. What's more, internal SIP users can be called from landlines by using standard telephone numbers. In this way, home offices and smaller sites can be cost-effectively equipped with a standardized telephony system based on SIP.

Model variants

This manual applies for all LANCOM Business-VoIP-Routers (1722, 1723, 1724 and 1823 VoIP).

Model restrictions

Passages applying only to certain models are identified either in the text itself or by a comment in the margin.

Otherwise the documentation refers to all models collectively as LANCOM Business-VoIP-Routers.

Components of the documentation

The documentation of your device consists of the following parts: The installation guide, the user manual and the reference manual. This additional documentation of the PBX functions is dedicated to the setup and operation of the LANCOM Business-VoIP-Routers.

All other information including the technical specifications, Internet-access configuration or technical background information is available from the user manual for your model or from the reference manual on the supplied data medium (CD/DVD).

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Contents

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1 Introduction

LANCOM Business-VoIP-Routers can provide small companies or subsidiaries with all of the functions of a classical private branch exchange (PBX).

- Connection of telephones and fax machines (ISDN or analog)
- Internal telephone calls between all connected subscribers
- Telephony functions such as call hold, swap, connect or call transfer (redirect calls)
- Hunt group function with flexible call distribution and cascading of hunt groups

Along with these functions familiar from ISDN PBXs, the LANCOM Business-VoIP-Router can also utilize all of the advantages of modern VoIP infrastructure:

- Choice of telephony via analog, ISDN or SIP (e.g. Internet telephony)
- Integration of SIP telephones and softphones
- Telephony between sites by using SIP via VPN
- Automatic directing of calls with intelligent call routing
- Configuration of dispersed PBXs by one central IT department

This documentation assumes that the basic configuration for your LANCOM device (Internet access, security settings) has been carried out already. The manual on PBX functions of LANCOM Business-VoIP-Routers deals with the following subjects:

1. Making calls with telephones and softphones
2. Connecting the LANCOM Business-VoIP-Router to the telephone network
3. Connecting terminal equipment to the LANCOM Business-VoIP-Router
4. Configuring the PBX functions
5. Activation of the VoIP-32 Option

This description is limited to a description of operating the LANCOM Business-VoIP-Router as a "stand-alone" PBX. For information on other LANCOM Business-VoIP-Router issues such as the combination with existing ISDN PBX systems, connection to an upstream SIP PBX, configuring SIP trunks, or similar matters, please refer to your product's user manual and/or the reference manual.

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2 Hardware installation

This chapter explains how to connect the LANCOM Business-VoIP-Router to the telephone network and how to connect terminal equipment to the LANCOM Business-VoIP-Router.

The connections described relate to the operation of the LANCOM Business-VoIP-Router as a "stand-alone" PBX. The devices can be connected in different manners, and information about these options can be found in the appropriate user manual and/or in the LCOS reference manual.

LANCOM Business-VoIP-Routers feature an ISDN interface and a number of LAN connectors, making them suitable for a simple SIP telephone infrastructure.

2.1 LANCOM 1723 VoIP and LANCOM 1823 VoIP

The VoIP routers LANCOM 1723 VoIP and LANCOM 1823 VoIP feature two ISDN interfaces, two LAN interfaces, two connectors for analog terminal devices and the option to connect an analog line to the exchange. This range of options is capable of supporting a sophisticated telephony infrastructure.

To operate in this mode, the DIP switches on the underside of the LANCOM 1723 VoIP and LANCOM 1823 VoIP devices must be in the following positions: DIP switches 1 to 8 in the up position, DIP switches 9 and 10 down. This is the ex-factory setting. Check the position of the DIP switches and set them correctly if necessary.

- Before altering the DIP switch settings, remove all cables from their sockets and switch the device off.
- Remove the see-through cover over the DIP switch.
- We suggest that you use a screwdriver to set the DIP switch to the desired position.
- Replace the see-through cover and reconnect the cables as described for your model in the user manual and as described in the following sections.

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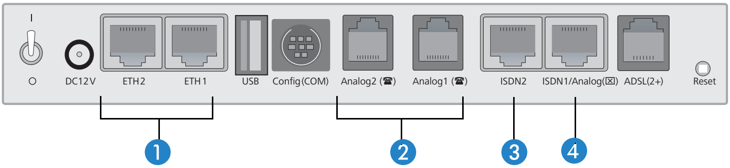
Example:
LANCOM 1723 VoIP



① **Connecting to the LAN** – First of all connect your LANCOM Business-VoIP-Router to the LAN. Plug in one end of the supplied network cable (green connectors) to a LAN connector on the device ❶, and the other end into an available network connector socket in your local network or a free socket on a switch or hub.

② **Connect the telephones and fax machines**

- Connect your SIP telephones to a network connection socket in your local network, or to a switch/hub that is connected to your LAN.
- Softphones are installed on the PCs, which are connected to the LAN.
- Analog devices (fax machines or DECT telephones) are connected to the LANCOM Business-VoIP-Router via the analog interfaces ❷ (RJ11 socket marked with ☎). For example, connect a DECT telephone with answering machine to Analog 1 and a fax machine to Analog 2.

  If your terminal equipment features a TAE-F or TAE-N connector, please use the supplied adapter cable (RJ11 plug to TAE-N/F socket).

  If necessary, further analog equipment can be connected to the LAN by means of an Analog Terminal Adapter (ATA).
- ISDN telephones can be connected to the ISDN 2 interface ❸ either directly or via the ISDN bus of an ISDN PBX.

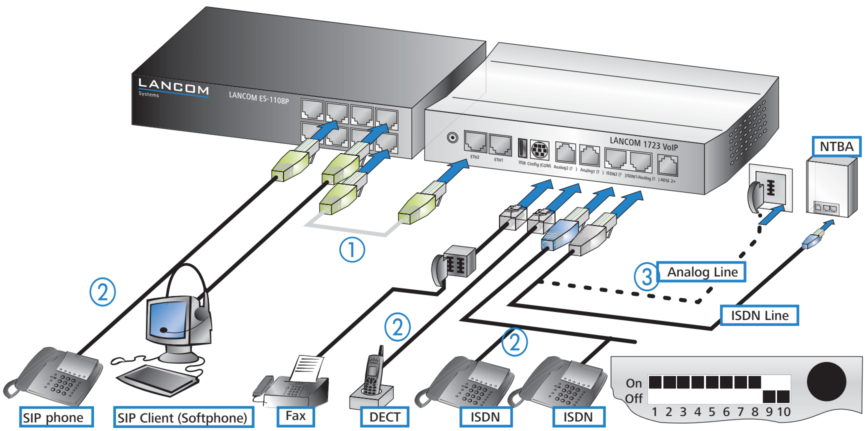
For the models LANCOM 1723 VoIP and LANCOM 1823 VoIP, **never** use the interface ISDN 2 ❸ to connect to the ISDN network (exchange)!

③ **Connecting to the public services telephone network**

The models LANCOM 1723 VoIP and LANCOM 1823 VoIP can be connected either to the ISDN network **or** to the analog telephone network:

- **Connecting to the ISDN** – to connect the LANCOM Business-VoIP-Router to the ISDN, plug in one end of the supplied ISDN cable (light-blue connectors) to the combined ISDN/analog interface 4. Plug in the other end of the ISDN cable into an ISDN/S_0 point-to-point line connector or point-to-multipoint line connector.
- **Connecting to the analog telephone network**—to connect the LANCOM Business-VoIP-Router to the analog telephone network, plug the end of the supplied analog connector cable marked in yellow (RJ45) into the combined ISDN/analog interface 4. The other end of the analog connector cable (RJ11) is to be plugged into an analog exchange line (e.g. a splitter). If the exchange line has a TAE-N/F socket, you can use the supplied adapter (RJ11 plug to TAE plug).

Connections LANCOM 1723 VoIP



2.2 LANCOM 1722 VoIP

The VoIP routers LANCOM 1722 VoIP feature two ISDN and four LAN interfaces, making them ideal for establishing SIP telephone infrastructures that feature four parallel ISDN channels for incoming and outgoing calls to the public telephone network.

To operate in this mode, all 10 of the DIP switches on the underside of the LANCOM 1722 VoIP must be in the down position! Set the DIP

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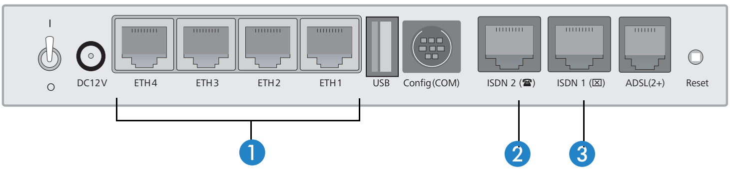
switches 1 to 8—supplied ex-factory in the up position—to the down position.

- Before altering the DIP switch settings, remove all cables from their sockets and switch the device off.
- Remove the see-through cover over the DIP switch.
- We suggest that you use a screwdriver to set the DIP switch to the desired position.
- Replace the see-through cover and reconnect the cables as described for your model in the user manual and as described in the following sections.

LANCOM 1722 VoIP



① **Connecting to the LAN** – First of all connect your LANCOM 1722 VoIP to the LAN. Plug in one end of the supplied network cable (green connectors) to a LAN connector on the device ❶, and the other end into an available network connector socket in your local network or a free socket on a switch or hub.

② **Connect the telephones and fax machines**

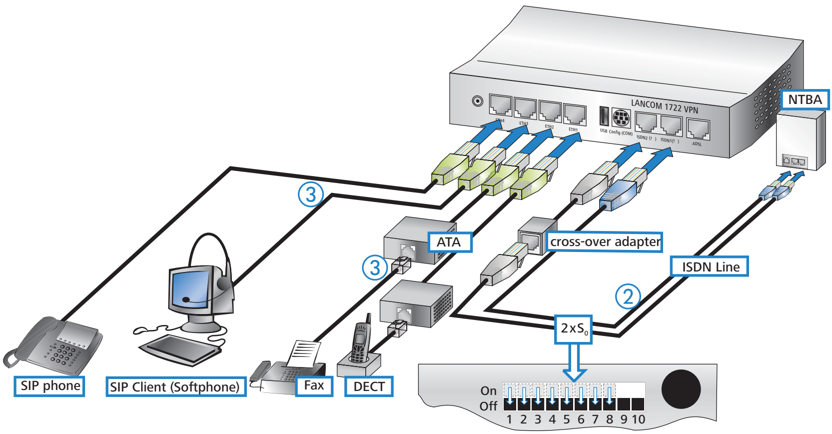
- Connect your SIP telephone to a free LAN interface on the LANCOM 1722 VoIP or to a switch/hub connected to your LAN.
- Softphones are installed on the PCs, which are connected to the LAN.
- Analog terminal equipment such as fax machines or DECT telephones can be connected to the LAN by using an Analog Terminal Adapter (ATA).
- In this example we will not discuss the connection of ISDN telephones because both ISDN interfaces on the LANCOM 1722 VoIP are used to connect to the public telephone network.

③ **Connecting to the ISDN network**

To be able to use all four ISDN channels for calls via the public telephone network, use the ISDN cross-over adapter (supplied) to plug a separate ISDN connector cable into the ISDN 1 interface ❷. Plug in the other end of this cable into an ISDN/S_0 point-to-point line connector or point-to-multipoint line connector.

Plug one end of the supplied ISDN connector cable (light-blue connectors) into the ISDN 1 interface 3. Plug in the other end of the ISDN cable into an ISDN/S_0 point-to-point line connector or point-to-multipoint line connector.

Connections LANCOM 1722 VoIP



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2.3 LANCOM 1724 VoIP

The VoIP routers LANCOM 1724 VoIP feature two LAN and four ISDN interfaces, making them ideal for establishing larger SIP telephone infrastructures that feature up to eight parallel ISDN channels for incoming and outgoing calls to the public telephone network.

To operate in this mode, all 10 of the DIP switches on the underside of the LANCOM 1724 VoIP must be in the down position! Set the DIP switches 1 to 4—supplied ex-factory in the up position—to the down position.

- Before altering the DIP switch settings, remove all cables from their sockets and switch the device off.
- Remove the see-through cover over the DIP switch.

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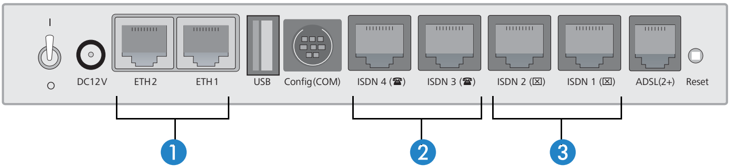
- ☐ We suggest that you use a screwdriver to set the DIP switch to the desired position.
- ☐ Replace the see-through cover and reconnect the cables as described for your model in the user manual and as described in the following sections.

LANCOM 1724 VoIP



① **Connecting to the LAN** – First of all connect your LANCOM 1724 VoIP to the LAN. Plug in one end of the supplied network cable (green connectors) to a LAN connector on the device ❶, and the other end into an available network connector socket in your local network or a free socket on a switch or hub.

② **Connecting to the ISDN network**

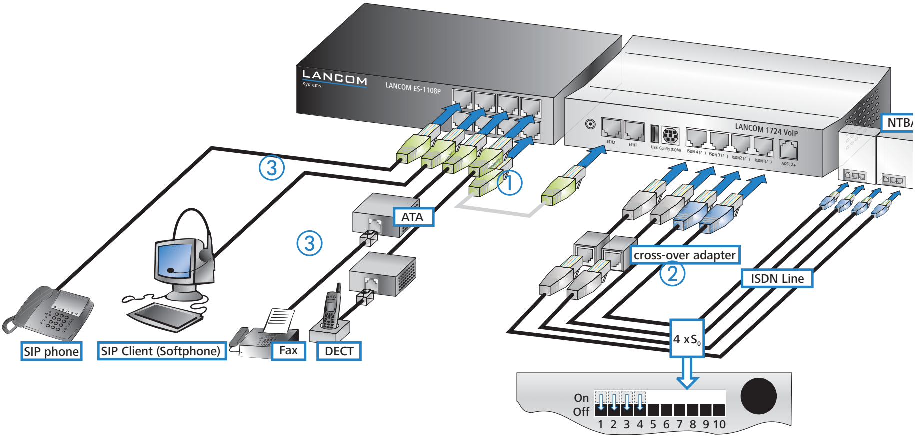
To be able to use all eight ISDN channels for calls via the public telephone network, use the ISDN cross-over adapters (supplied) to plug two separate ISDN connector cables into the ISDN 3 and ISDN 4 interfaces ❷. Plug in the other end of this cable into an ISDN/S_0 point-to-point line connector or point-to-multipoint line connector.

Plug one end of each supplied ISDN connector cable (light-blue connectors) into the ISDN 1 interface and ISDN 2 interface ❸. Plug in the other end of each ISDN cable into an ISDN/S_0 point-to-point line connector or point-to-multipoint line connector.

③ **Connect the telephones and fax machines**

- ☐ Connect your SIP telephones to a network connection socket in your local network, or to a switch/hub that is connected to your LAN.
- ☐ Softphones are installed on the PCs, which are connected to the LAN.
- ☐ Analog terminal equipment such as fax machines or DECT telephones can be connected to the LAN by using an Analog Terminal Adapter (ATA).
- ☐ In this example we will not discuss the connection of ISDN telephones because all four ISDN interfaces on the LANCOM 1724 VoIP are used to connect to the public telephone network.

Connections LANCOM 1724 VoIP



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3 Configuring the VoIP functions

Prerequisites for the configuration of the VoIP functions in a LANCOM Business-VoIP-Router are suitable basic settings and a functional Internet connection. To this end, please ensure that you use the Wizards in LANconfig to configure the basic settings, the Internet connection and the security settings before you configure VoIP.

3.1 This is how you draw up a dialing plan

The planning of telephone infrastructure can be greatly helped by drawing up precise details on the connections, their telephone numbers and any hunt groups that may be required, as this information is important to the configuration. To help you with this, you should fill out the table (below) before you begin with the configuration.

3.1.1 PBX users

Users are all of the devices for tele(phonic) communications, i.e. ISDN and analog telephones, SIP telephones, computers with software telephones, or fax machines. As a rule, every user receives its own internal telephone number that can be accessed from within the company. If necessary, multiple devices can share the same telephone number ('Multi-login' →Page 36).

Use a uniform number format so as to avoid problems with the processing of internal telephone numbers, e.g. use two- or three-digit numbers only.

The external telephone number is used to reach the user from the PSTN (public services telephone network). At ISDN point-to-multipoint connections these numbers are referred to as MSNs (Multiple Subscriber Numbers). At an ISDN point-to-point connection you enter the direct-access telephone number (DDI, Direct Dialing-In). You should then enter the type of device and connection.

Please observe the maximum number of users permissible for your LANCOM model.

The following table provides an overview of the allocation of users to hunt groups ('Hunt groups' →Page 16), which suffices for many applications.

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Internal Telephone number	User or group name	Ext. tel. number/ MSN/DDI	Device type (Telephone, Fax)	Connection (SIP, ISDN, analog)
10 (hunt group)				
11				
12				
13				
14				
15				
16				
17				
18				
19				
20 (hunt group)				
21				
22				
23				
24				
25				
26				
27				
28				
29				
30 (hunt group)				
31				
32				
33				
34				
35				
36				
37				
38				
39				

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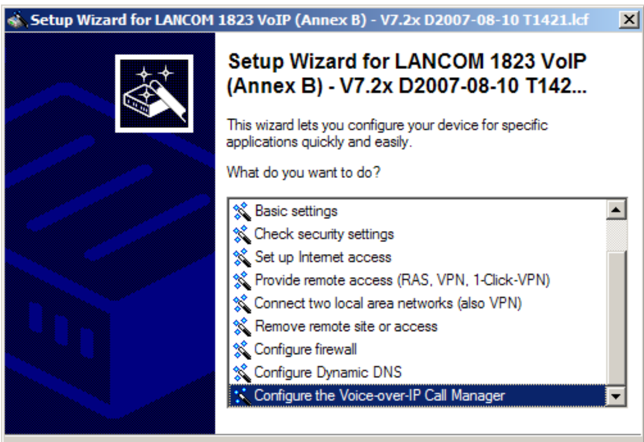
3.1.2 Hunt groups

A hunt group collects two or more users who/which should be reached under a single telephone number. For example, a hunt group can be used to dial all of the staff within a department ('Hunt groups with call distribution' →Page 34). Apart from its members, a hunt group can also be set with the manner of call distribution, a time period after which (unanswered) calls are forwarded, and a number as the target for call forwarding.

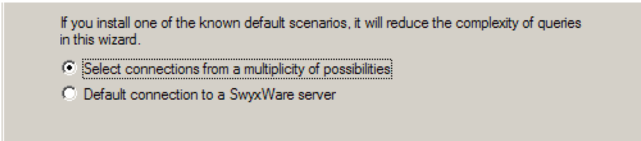
Hunt group	Comment	Members	Distribution	Forwarding time	Forwarding target
10					
20					
30					

3.2 This is how you configure the LANCOM Business-VoIP-Router as a PBX

① Under LANconfig, start the setup wizard for configuring the VoIP Call Manager.

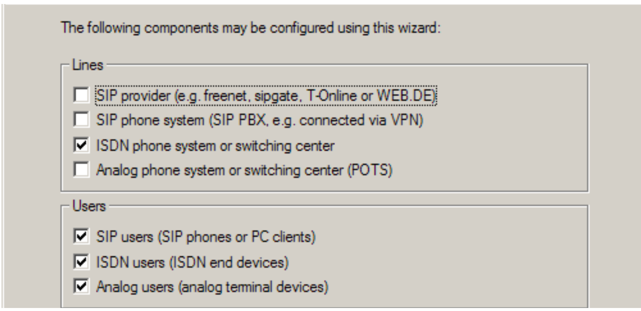


② Choose the option 'Select connections from a multiplicity of possibilities'.

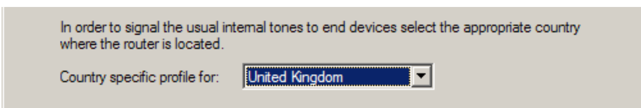


③ Under 'Lines', activate the exchange line over which you telephone via the PSTN (public services telephone network): ISDN or analog connection. If your model features both analog and ISDN interfaces, please observe that you can only activate one of these two options.

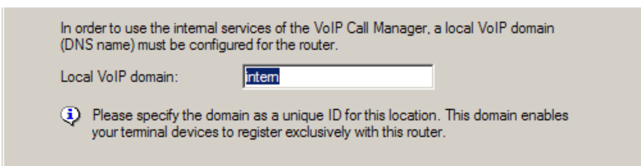
You then select the type of telephones that are to be used: SIP, ISDN or analog users (analog users are not available on all models).



④ Set the country in which the LANCOM Business-VoIP-Router is to be operated.



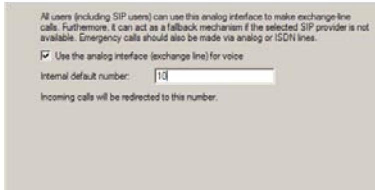
⑤ The local VoIP domain can be left as 'internal', unless a different domain is required for operation with another system (in most cases this is unnecessary).



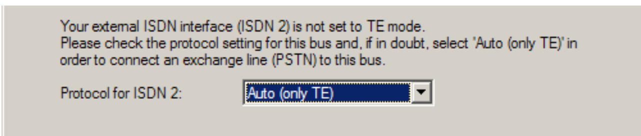
⑥ You then activate your ISDN or analog connection for voice communications, so that you can use this connection to telephone to the public telephone network.

With an ISDN connection, you can select which of the available ISDN interfaces is to be used for external telephone calls.

For an analog connection, you can define the internal telephone number (or hunt group) that incoming calls are to be forwarded to.

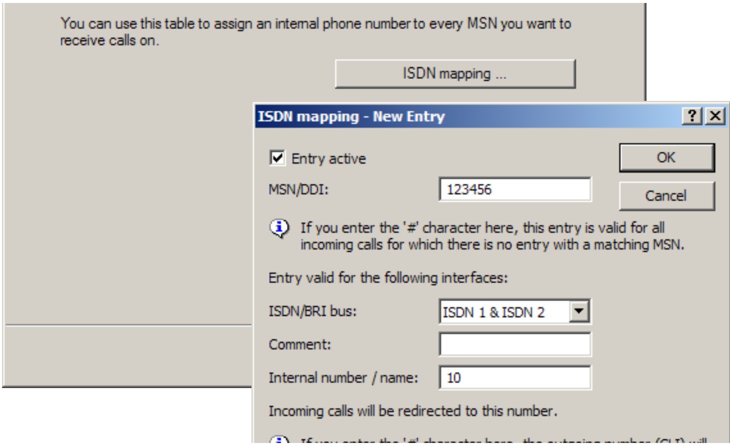


⑦ Depending on the model, the wizard may ask you to set the protocol required by the ISDN interfaces for external telephony. Set the protocol for each interface to 'Auto (TE only)'.

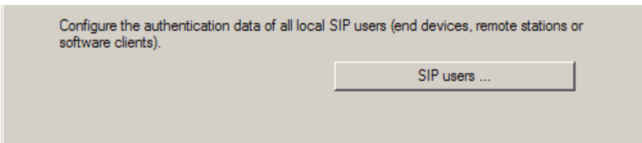


⑧ If you use an ISDN connection to telephone via the public telephone network, you can give every external telephone number (MSN or DDI) a corresponding internal telephone number. Use the allocation in your dialing plan to help you with this. At the same time, if applicable, you select the ISDN interface over which these external telephone numbers receive calls. If multiple ISDN interfaces to the exchange line have been configured as point-to-point connections, then you also have to select each DDI that corresponds to these interfaces.

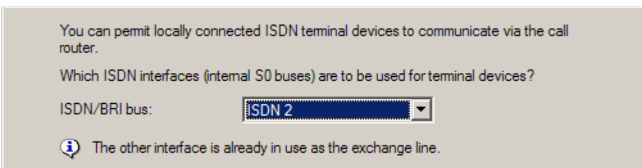
In case you wish to use an MSN or DDI as a central telephone number for a group of subscribers (hunt group), then with the help of your dialing plan you can enter the internal telephone number for the group here, e.g. '10' to reach the subscribers '11', '12', '13', etc.



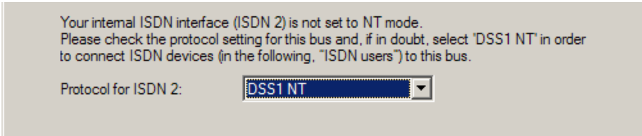
⑨ You can skip the configuration of SIP users, assuming that you **do not** require the SIP users to register with user name and password.



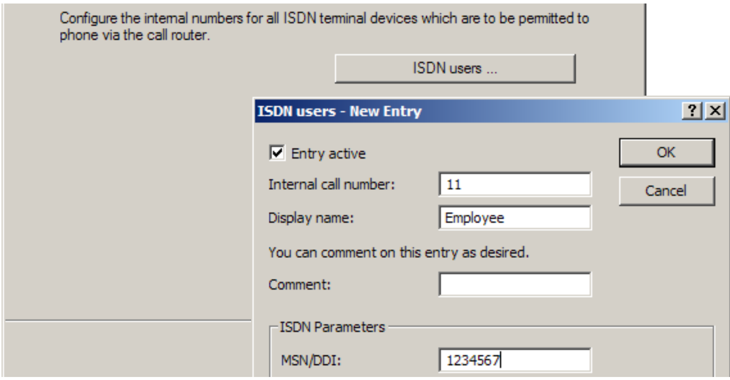
⑩ If your model has free ISDN interfaces for connecting ISDN terminal equipment, you can select which of these interfaces are to be used for connecting internal terminal devices.



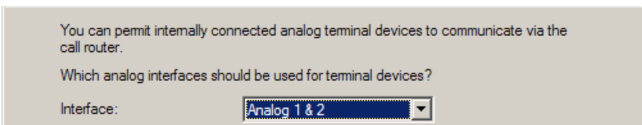
⑪ Depending on the model, the wizard may ask you to set the protocol required by the ISDN interfaces to connect these internal terminal devices. The protocol should be set to 'DSS1 NT' for point-to-multipoint connections (if, for example, you wish to connect individual telephones directly to the corresponding interface) or to 'DSS1 NT point to point' for point-to-point connections.



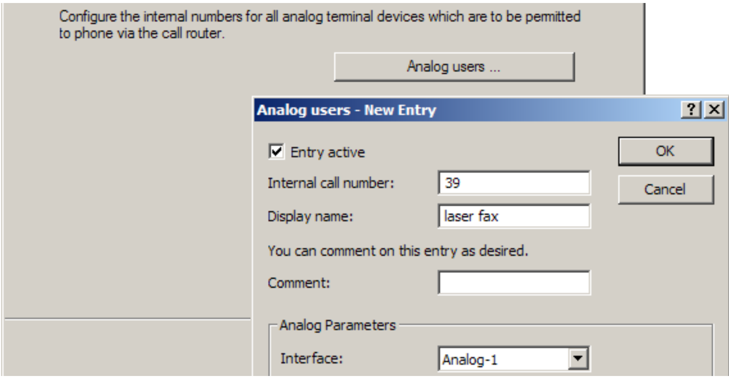
⑫ For each ISDN terminal device that is connected, you can define the corresponding internal telephone number and the associated external MSN or DDI. At the same time you select the ISDN interface that the terminal device is connected to.



⑬ If your model has analog interfaces for connecting analog terminal equipment, you can select which of these interfaces are to be used for connecting internal terminal devices.

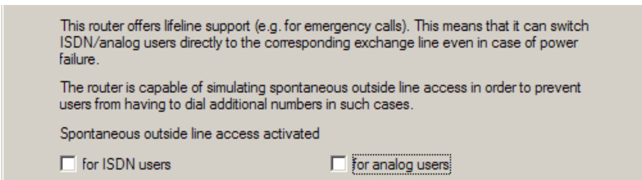


⑭ For each analog terminal device that is connected you can define the corresponding internal telephone number. At the same time you select the analog interface that the terminal device is connected to. Please ensure that you also select the device type: 'Phone', 'Fax' or 'Phone and/or fax' (in case you have a combination device or if phone and fax are connected to an analog adapter).



⑮ In the following stages the call router is set up to control the processing of incoming and outgoing calls. First of all you can select whether to precede external calls with a zero. Spontaneous outside line access means that **no** leading zero is required. This can be set separately for ISDN and analog users.

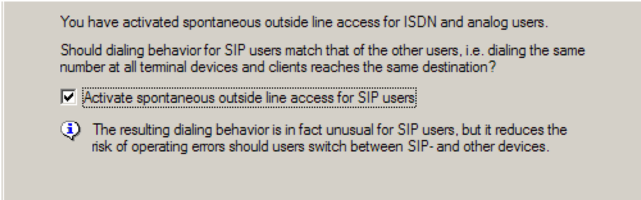
Typically you will want to make calls to other people in your company simply by entering their extension number directly. This is only possible if you do not use spontaneous outside line access.



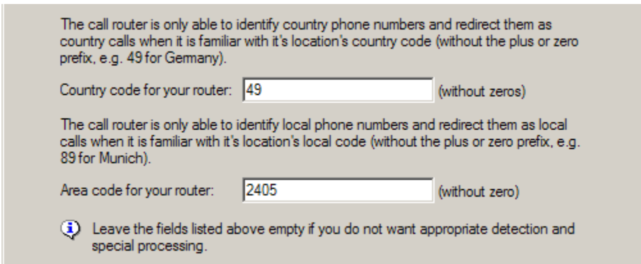
When using the LANCOM Business-VoIP-Router as a PBX, we recommend that you deactivate spontaneous outside line access. This ensures that the leading zero for external calls has to be entered, a procedure that is familiar to users of conventional PBXs.

⑯ If you do decide to use spontaneous outside line access, the wizard also provides this for SIP users so that all telephones are operated in the same way.

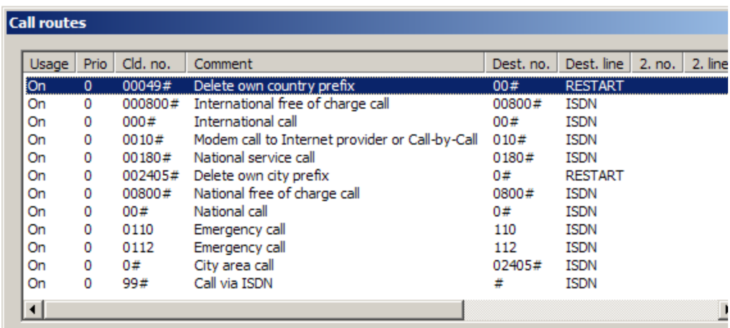
When spontaneous outside line access is activated, the wizard generates dialing rules to enable you to call internal numbers by entering a leading asterisk, e.g. '*11'.



⑰ Entering your national and area code enables the Call Router to handle local or national calls in certain ways.



⑱ Once the configuration is completed, the wizard displays an overview of the call routes that are to be generated. Further information on the function of call routes and their significance is available in the LCOS reference manual.



Call routes

Usage	Prio	Cld. no.	Comment	Dest. no.	Dest. line	2. no.	2. line
On	0	00049#	Delete own country prefix	00#	RESTART		
On	0	000800#	International free of charge call	00800#	ISDN		
On	0	000#	International call	00#	ISDN		
On	0	0010#	Modem call to Internet provider or Call-by-Call	010#	ISDN		
On	0	00180#	National service call	0180#	ISDN		
On	0	002405#	Delete own city prefix	0#	RESTART		
On	0	00800#	National free of charge call	0800#	ISDN		
On	0	00#	National call	0#	ISDN		
On	0	0110	Emergency call	110	ISDN		
On	0	0112	Emergency call	112	ISDN		
On	0	0#	City area call	02405#	ISDN		
On	0	99#	Call via ISDN	#	ISDN		

⑲ Well done! You have configured the LANCOM VoIP Router as the local PBX. Please observe the following information to set up the various terminal devices.

If your dialing plan foresees the use of hunt groups, these still have to be entered into the configuration manually ('Hunt groups with call distribution' →Page 34).

3.3 This is how you configure your telephones and terminal equipment

Apart from configuring the LANCOM Business-VoIP-Router itself, some setting up of the terminal equipment is also necessary to ensure that it remains fully functional in the telephone network.

3.3.1 Analog telephones

Analog telephones use the Flash key (F key) to control certain functions. Pressing the F (or Flash-) key causes the line to be interrupted briefly. The PBX detects this flash and interprets any following tones not as numbers but as control signals intended for the PBX.

The duration of this interruption plays a significant role in the triggering of control sequences. LANCOM Business-VoIP-Routers expect a "short" flash of 80 to 150 ms. Many devices are preset to use the "long" flash of 170 to 310 ms.

For this reason you should set your analog telephone to use the short flash. Many devices allow this value to be adjusted in a menu under **Settings** ▶ **PBX** ▶ **Flash**. If your model allows an explicit flash duration to be defined, set the flash-time value to 100 ms.

If necessary, please refer to the documentation for your device for information on these settings.

3.3.2 ISDN telephones

The use of ISDN telephones requires the configuration of the MSN in the terminal device. To make use of PBX functions in the LANCOM Business-VoIP-Router, the parameters "ECT" and "Keypad" have to be checked and/or set.

If necessary, please refer to the documentation for your device for information on these settings.

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MSN setting

By entering an MSN into an ISDN telephone, you define the telephone number that the phone is to react to and the telephone number that is transmitted to the user being called.

The MSN for your ISDN telephone should be the telephone number that you have entered for the ISDN user under 'MSN/DDI'. In many devices, this setting is to be found in the menu under **Telephone settings ▶ MSN**.

Not to be entered as the MSN is the internal telephone number! Using the external MSN/DDI enables the ISDN telephone to continue operating even in case the LANCOM Business-VoIP-Router should fail (’Life-line support for ISDN telephones’ →Page 45).

ECT setting

ECT (Explicit Call Transfer) is the spontaneous connection of two active calls by the user: If you are conducting two conversations, you can transfer the two callers so that they can then communicate with each other (’Spontaneous call management by the user’ →Page 29).

To use this switching option in combination with the LANCOM Business-VoIP-Router, you should activate automatic ECT in your ISDN telephone. Many devices allow this value to be adjusted in a menu under **Telephone settings ▶ PBX ▶ Functions▶ Auto. ECT**.

Keypad setting

For ISDN telephones, control sequences use the # and * characters (known as "keypads") to to activate/deactivate features at the exchange such as automatic call forwarding (call redirection) or callback on busy. Pressing keys during a call normally generates DTMF tones, which can be used to control remote computers, for example.

To prevent control sequences for features from being transmitted as DTMF tones, activate automatic keypad transmission for your ISDN telephone. Many devices allow this value to be adjusted in a menu under **Telephone settings ▶ PBX ▶ Functions▶ Keypad**.

3.3.3 SIP telephones

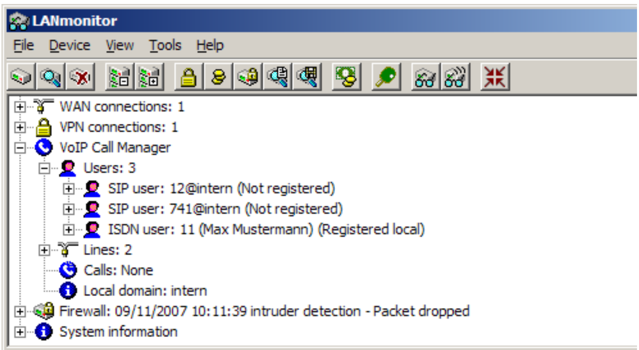
Similar to ISDN telephones, SIP telephones also require a setting for the telephone number that the device is to react to. If the LANCOM business VoIP router operates as the DHCP and DNS server in the LAN and local authentication is switched off (default setting), then a SIP telephone that finds the SIP realm by DNS only requires this number in order to register.

① It is very easy to set up the telephone number on a telephone that supports LANCOM Easy Setup: After starting the SIP telephone in its factory settings, it may be necessary to make some basic settings such as language, time, time zone, etc.

② No account has been configured yet, so the device will request that you enter details for the first SIP account. All you have to do is to enter the internal telephone number and confirm. If the device supports LANCOM Easy Setup, then the setup is complete.

If you have set up the VoIP Call Manager to force the authentication of local subscribers, then the SIP telephone will request the entry of the password for the associated account.

The SIP phone now tries to register at the LANCOM Business-VoIP-Router with the number entered. If registration is successful, the telephone displays its internal telephone number. You can also check the registration in LANmonitor: All registered users have an entry which displays their current status.



If necessary, please refer to the documentation for your device for information on these settings.

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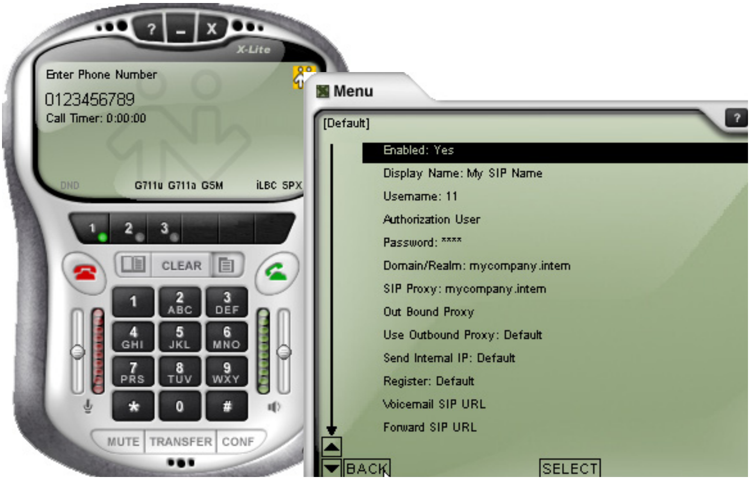
3.3.4 Software telephones (SIP softphones)

Enter the registration information for the first SIP user in the softphone (example for X-Lite).

① From the menu, select **System Settings ▶ SIP Proxy** and choose an entry for one of the possible lines, e.g. 'Default'.



② Enter the following values:

- ▷ Display name: Name of the user as it is to be displayed at the remote site.
- ▷ Domain/realm: Internal VoIP domain for the LANCOM.
- ▷ SIP proxy: Internal VoIP domain for the LANCOM.
- ▷ User name: Internal number for the user.

If necessary, please refer to the documentation for your softphone for information on these settings.

3.3.5 Analog Terminal Adapters (ATAs)

Analog terminal adapters are used to connect analog terminal equipment such as DECT telephones or fax machines to the LAN. This method enables these devices to be integrated into the telephone infrastructure just like SIP devices. The following parameters have to be set to operate an ATA at a LANCOM Business-VoIP-Router that functions as a PBX:

- The telephone number that the device is to react to.

- Registry information such as registrar, SIP proxy or realm (the naming of these parameters can vary from device to device):
  - Here you enter the internal VoIP domain of the LANCOM Business-VoIP-Router (default: 'internal') if the ATA is to retrieve this information from the LANCOM Business-VoIP-Router operating as DHCP and DNS server.
  - Otherwise you should enter the local IP address of the LANCOM Business-VoIP-Router.
- If you connect a fax machine, select the fax transmission method as desired or as supported by the ATA.
  - Where fax signals are to be transmitted like voice data over a VoIP connection, this is referred to as "fax over VoIP". Fax transmission requires the use of the compression codec G.711. Accordingly, set this codec in you ATA.
  - As an alternative, fax messages can be transmitted using the T.38 standard, meaning that they are not transmitted as voice signals via VoIP, but rather with a special protocol known as the IFP (Internet Facsimilé Protocol). If your ATA supports this method, set fax transmission to T.38 ("Fax over IP" or FoIP).

Further information about faxing with T.38 is available in the LCOS reference manual.

Now you have found out a great deal about setting up a LANCOM Business-VoIP-Router as a PBX in combination with a variety of different terminal equipment.

The following chapters inform you about how you can set up and operate the PBX functions available with this high-performance installation.

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4 PBX functions in the LANCOM Business-VoIP-Router

LANCOM Business-VoIP-Routers can provide small companies or subsidiaries with all of the functions of a classical private branch exchange (PBX).

- Telephony functions such as call hold, swap, connect or call transfer (redirect calls)
- Hunt group function with flexible call distribution and cascading of hunt groups
- Multiple logins to use various telephones under one telephone number

4.1 Call forwarding

Call forwarding means that a call that has already been placed is redirected to a new destination either spontaneously by the user ("call transfer") or by automatic call forwarding ("redirect call") as set up in advance. Call forwarding with SIP-based VoIP telephony uses a different technology to that used formerly, and yet the implementation of this function in the LANCOM Business-VoIP-Router means that its operation is almost identical for all types of terminal device.

Call charges for external call forwarding

The transfer of a call from an external caller to a third party who is also external carries the risk that charges will arise for the ongoing call, even though the initiating subscriber has ended the call.

When forwarding calls from external callers to another external subscriber (e.g. when forwarding an office number to a mobile phone) both of the ISDN lines are engaged when the external calls are directed over a single ISDN line. If your device is connected to the public telephone network via just two ISDN channels (one ISDN interface), then no further calls can be made. Alternatively you can try 'Setting up call forwarding in the telephone exchange' →Page 43.

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4.1.1 Spontaneous call management by the user

Functions for spontaneous call management

Calls can be managed on an individual basis and the LANCOM Business-VoIP-Router supports the services known from the ISDN network:

- With **call hold** the user can place an active call into a wait state. In this state, the user can for example make a call to another person.
- Establishing an additional call while a call is on hold is referred to as **consulting**. This call can be ended again and the conversation with the call on hold continued.
- With **swap call**, the user can switch to and fro between two connections. The user is only connected with one caller at a time, while the other caller is put on hold.
- With **call transfer** ("connect call") the user switches an active call over to another call which is on hold. The two callers are then connected and the user is no longer involved in the call. A subscriber transferring a call can either directly hand over an active call to a third subscriber (unattended call transfer), or a separate call can be made to the third subscriber to communicate the call and then transfer it (attended call transfer).

Using spontaneous call management with various telephones

SIP telephones and SIP softphones generally feature special keys or menu entries to manage calls. Depending on the model or program, different terms may be used the the functions are as follows:

- HOLD: Places an active call into a wait mode or swaps between two active calls. On ISDN and analog telephones this function is often referred to as the F-key/Flash/Call hold.
- HANG UP: End the current call.
- SWAP: Swap between two active calls (depending on the ISDN telephone, this may be initiated by a display-menu entry, a special key, or the "F" key).
- CONNECT: Initiates the call's transfer (can be triggered by "hanging-up" with an active call and a call on hold)*.

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These functions can be used to manage calls as follows:

Holding/consulting and continuing with calls	SIP	ISDN	Analog
To place a call on hold, press the Flash/Call hold key (or 'F' on analog phones). The caller can no longer hear you and you can initiate a second call by dialing a telephone number (consulting).	HOLD	HOLD or F	R
To continue with a call which is on hold, press the Flash/Call hold key again (or 'F 2').	HOLD	HOLD or F	R 2 or R after a short delay
If the consultation call has not yet been picked up, you can stop the consulting by hanging up the handset on a SIP or ISDN telephone*. You can stop the consultation call with the appropriate menu function of the telephone (e.g. 'Cancel') or 'F 1' (analog).*	HANG UP	HANG UP	HANG UP

Swap	SIP	ISDN	Analog
To open a second line during a call, first press the Flash/Call hold key (or 'F' on analog phones). The other caller can no longer hear you.	HOLD	HOLD or F	R
Dial the number for the second caller while the first call is on hold. If you cannot reach the second caller, you can return to the call which is on hold by pressing the hold key (or 'F').	123456789	123456789	123456789
As soon as two simultaneous connections are open, you can use the hold key (or swap key for ISDN phones, 'R' and '2' for analog phones) to swap to-and-fro between the two connections. You will be connected to one of the other callers; the other caller is placed on hold.	HOLD	SWAP	R 2 or R after a short delay
To end an active call, hang up the handset on SIP or ISDN telephones, and on analog phones press 'F 1'. The call which is on hold is not automatically reactivated, but it will be signaled (ringing phone) for a period of 15 seconds.	END or HANG UP*	END or HANG UP*	F 1

Call transfer, consult	SIP	ISDN	Analog
To open a second line during a call, first press the Flash/Call hold key (or 'F' on analog phones). The other caller can no longer hear you.	HOLD	HOLD or F	R

Call transfer, consult	SIP	ISDN	Analog
Dial the number for the second caller while the first call is on hold. If you cannot reach the second caller, you can return to the call which is on hold by pressing the hold key.	123456789	123456789	123456789
As soon as you have established two simultaneous connections you can connect the two callers with the connect key (or 'F' and '4' on analog phones) or by hanging up the handset.* Optionally you can switch between the two lines as often as you like before transferring. Call transfer always connects the active call and the call on hold.	CONNECT or HANG UP*	CONNECT or HANG UP*	R 4 or HANG UP
You have no more active calls. You can replace the handset.	HANG UP	HANG UP	HANG UP

Call transfer, blind	SIP	ISDN	Analog
To open a second line during a call, first press the Flash/Call hold key. The other caller can no longer hear you.	HOLD	HOLD	HOLD
Dial the number for the second caller while the first call is on hold.	123456789	123456789	123456789
Press the connect key (or 'F' and '4' on analog phones) or hang-up the handset before the second connection has been established.* The two callers will now be connected "in the background".	CONNECT or HANG UP*	CONNECT or HANG UP*	R 4 or HANG UP
You have no more active calls. You can replace the handset.	HANG UP	HANG UP	HANG UP

*In some cases, SIP or ISDN telephones can be configured so that hanging-up the handset either causes the consultation or active call or be terminated, or a call transfer is triggered ("Connect").

4.1.2 Configure permanent call forwarding

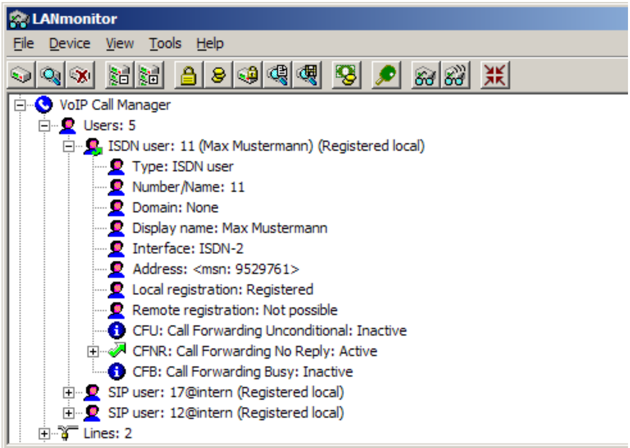
Along with spontaneous call transfers as controlled by a subscriber during a call ("connect call", it is often useful to set up a permanent call forwarding ("redirect calls"). For example, a call should be forwarded when a line is busy, if there is no answer within a certain period, or in case of absence (e.g. vacation).

There are two possibilities for configuring permanent call forwarding.

- Via the telephone or terminal device itself with the aid of control characters
- In the configuration of the LANCOM Business-VoIP-Router by means of the management tools (LANconfig, WEBconfig or telnet)

If permanent call forwarding is activated by both methods, then the behavior of the call forwarding follows the last respective action.

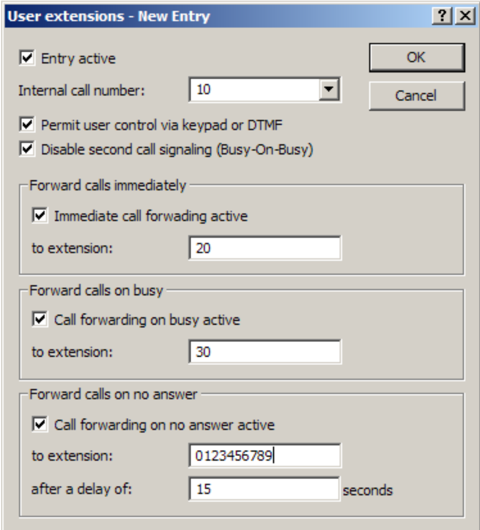
The setting for call-forwarding is displayed for each user in LANmonitor with the symbol shown in the margin. The values set for forwarding target and delay time can be displayed in the details for each user.



Call forwarding that is configured directly in the exchange ('Setting up call forwarding in the telephone exchange' →Page 43) is also not shown in LANmonitor.

Configuring call forwarding in the LANCOM Business-VoIP-Router

To configure call forwarding in LANconfig, go to**VoIP Call Manager ▶ Users ▶ User settings**:



① Select the internal telephone number that call forwarding applies to.

Call forwarding can be set up for all local users (SIP, ISDN or analog).

② You can define whether the user at the selected internal telephone number can alter the settings for call forwarding by means of the terminal device's keypad.

③ Under "Forward calls immediately" enter the target telephone number for immediate call forwarding when a call is made to the intended internal number.

④ Under "Forward calls on busy" enter the target telephone number for call forwarding if the called internal number is busy.

⑤ Under "Forward calls on no answer" enter the target number for calls to be forwarded to in case there is no answer, and enter the delay defining the how long the telephone rings before the call is forwarded.

Calls can be forwarded to local users, hunt groups, or external telephone numbers. Enter external telephone numbers as

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call-forwarding targets just as you would dial the subscriber from the local telephone network. If you have deactivated spontaneous outside line access and you have to dial a zero for outside calls, place a leading zero before the call forwarding target here as well.

Configuring call forwarding with the telephone

Configuring call forwarding in the LANCOM Business-VoIP-Router from the telephone uses the following combinations of characters:

Immediate call forwarding	SIP, ISDN, Analog
Switch on and define target for call forwarding	*21*TargetNo#
Switch off	#21#
Switch off temporarily, maintain call-forwarding target	#22#
Switch on again, maintain defined call-forwarding target	*22#

Call forwarding on busy	SIP, ISDN, Analog
Switch on and define target for call forwarding	*67*TargetNo#
Switch off	#67#

Call-forwarding on no reply	SIP, ISDN, Analog
Switch on and define target for call forwarding	*61*TargetNo#
Switch off	#61#

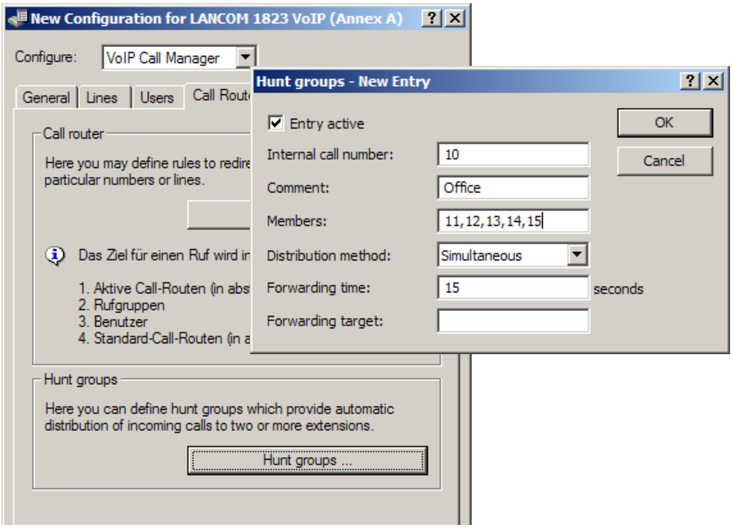
Some ISDN telephones feature special keys or menu entries to configure call forwarding, and these can be used as an alternative to the listed character strings. Refer to the documentation from the corresponding manufacturers.

If your telephone does not feature dedicated command keys, you can store these commands as speed-dial numbers and store them to dedicated keys.

4.2 Hunt groups with call distribution

Calls are normally intended for an individual or their telephone number. Occasionally it is not important to speak to a particular individual, but to anybody in a certain department or with a certain function. In this case, telephone infrastructure collects multiple users into hunt groups where they can all be reached under a single shared telephone number.

To configure call forwarding in LANconfig, go to **VoIP Call Manager ▶ Call routerr ▶ Hunt groups**:



① Enter the internal telephone number where the hunt group is to be reached.

② Enter the members of this hunt group with each entry separated by a comma. Members can be users, hunt groups or external telephone numbers.

Just like for call forwarding, members' call numbers are to be entered as their internal call numbers. The numbers also have to be entered in this format if you do decide to use spontaneous outside line access and internal numbers have to be dialed with a leading "*" symbol. For example, '*11' with spontaneous outside line access instead of '11' without.

③ Set the type of call distribution:

- Simultaneous: The call is signaled to all group members at once. If a member picks up the call within the call-forwarding time, the call is no longer signaled to other group members. If nobody accepts the call within the forwarding time, then the call is switched to its forwarding target.

- Sequential: The call is directed to one member of the group after the other. If a group member does not accept the call within the forwarding time, then the call is switched to the next member of the group. If nobody in the group accepts the call within the forwarding time, then the call is switched to its forwarding target.

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④ Set the forwarding time. If an incoming call is not picked up by a group member within the forwarding time, then the call is forwarded according to the distribution method selected:

- In case of simultaneous call distribution, the call is forwarded to the forwarding target.
- In case of sequential call distribution, the call is forwarded to the next group member in line. If the group member is the last one, then the call is redirected to its forwarding target.

⑤ Finally, enter the forwarding target. If none of the group members accepts the call within the forwarding time, then the call is switched to the forwarding target entered here. Forwarding targets can be users, hunt groups or external telephone numbers.

4.3 Multi-login

For subscribers using multiple terminal devices, e.g. a softphone on PC and a "normal" telephone on the desktop, multiple SIP, ISDN or analog telephones all using the same internal telephone number can log on to the LANCOM Business-VoIP-Router. Multi-login telephones behave like a single user in a hunt group set for 'simultaneous' call distribution (parallel call/twinning):

❶ Incoming calls are signaled **simultaneously at all** telephones with this internal number.

❷ As soon as a call is picked up at one of the telephones, signaling at the other devices stops.

❸ Other incoming calls are signaled at all telephones. If one of the telephones is 'busy', then the entire multi-login group is taken to be 'busy'.

❹ Outgoing calls can be made from every telephone without limitation.

5 For a multi-login group only one call forwarding setting (call redirection) can be configured. This applies to all telephones and can be set from any telephone.

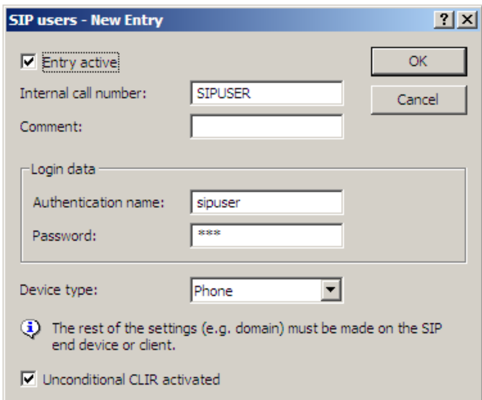
To use multi-login, multiple telephones can be set to have the same internal telephone number.

4.4 Calling Line Identification Restriction (CLIR)

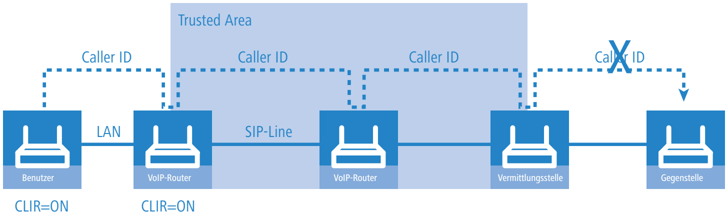
The CLIR function prevents the transmission of information about the calling party. This is usually activated directly at the terminal equipment, i. e. in the SIP phone or softphone, the subscribing user sets whether or not the calling number is to be displayed to the called party. Often, this behavior can be set up to occur generally or on a call-by-call basis.

If you activate CLIR in the LANCOM Business-VoIP-Router directly in the user settings for SIP, ISDN or analog users, the number will always be restricted, whatever settings the user makes in the terminal equipment.

CLIR is configured in LANconfig under **VoIP Call Manager ▶ Users ▶ SIP users**, **▶ ISDN users** or **▶ Analog users**:



In some cases it may be desirable for the caller ID to be retained even if it is restricted in the telephone or in the user settings. For instance, this is the case where a SIP provider requires the caller ID for billing purposes.



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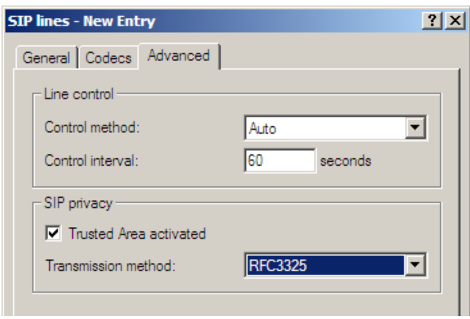
In this case the line to the SIP provider is flagged as belonging to a "trusted area". Within the trusted area, the caller ID is transmitted in a separate field (assuming that the privacy method is selected)—even if CLIR is activated in the telephone or in the LANCOM Business-VoIP-Router user settings. Only when the call leaves the trusted area (at the last exchange before the remote subscriber) is the caller ID removed as defined in the telephone or user settings. This means that the caller ID can be processed within the trusted area, but it will not be displayed by the remote device.

The privacy method is configured in LANconfig under **VoIP Call Manager ▶ Lines ▶ SIP lines** or **▶ SIP PBX lines ▶ Extended**:



5 Installing the LANCOM VoIP- Option

This chapter describes how the LANCOM VoIP-32 Option is installed to your LANCOM Router. Installation of the takes place in four steps:

① Checking the prerequisites for installation

② Online registration

③ Activation of the option

④ Checking the activation

To extend the number of SIP subscribers on LANCOM business VoIP routers to 32 you can activate the LANCOM VoIP-32 Option.

5.1 Requirements for installation

5.1.1 System requirements

Please ensure that you have met all of the requirements to successfully operate the LANCOM VoIP-Option:

- LANCOM Business-VoIP-Router with optional upgrade from 8 to 32 SIP subscribers.
- LCOS Version 7.22 or higher.

5.1.2 Package content

Please ensure that the Option package includes the following components:

- Proof of license with a printed license number
- Manual

5.1.3 Configuration computer with the Windows operating system

To install the LANCOM VoIP-Option you require a computer with a current Windows operating system. Alternatively, activation can be performed via WEBconfig.

The computer must have access to the LANCOM device that is to be configured. Access may be via the local network or even via remote access.

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5.1.4 Up-to-date LANconfig

The latest version of LANconfig and LANmonitor are available for download from the LANCOM Systems homepage under www.lancom.eu/download/. We recommend that you update these programs before continuing to the installation.

5.1.5 Up-to-date firmware in the LANCOM

The latest firmware updates are available for download from the LANCOM Systems web site in the customer portal myLANCOM. Select your device from the list and download the firmware onto your computer.

Detailed information about updating the firmware is available in the documentation for your LANCOM router.

5.2 Online registration

To activate the LANCOM VoIP-Option you need an activation code.

Please note: The activation code is not included in the package. It will be sent to you on online registration.

The LANCOM VoIP-Option is supplied with a proof of license. This has a license number printed on it. This license number gives you one opportunity to register with LANCOM Systems and to receive an activation code.

After successful online registration, the license number of your LANCOM VoIP-Option becomes invalid. The activation code that is supplied can only be used with the LANCOM as identified by serial number upon registration. Please ensure that you only want to install the LANCOM VoIP-Option on the corresponding device. It is not possible to change to another device at a later date!

Necessary registration information

Please have the following information at the ready for your online registration:

- Precise designation of the software option
- The license number (from the proof of license)
- Serial number of your LANCOM (to be found on the underside of the device)
- Your customer data (company, name, postal address, e-mail address).

Registration is anonymous and can be completed without specifying personal data. Any additional information may be of help to us in case of service and support. All information is of course treated with the strictest confidence.

Online entry of registration information

① Start a web browser and access the LANCOM Systems web site under www.lancom.eu/routeroptions.

② Enter the information as required and follow the subsequent instructions. If you submit an e-mail address you will receive confirmation of registration via e-mail. Online registration has been completed.

Make sure you store your activation code safely! You may need it at a later date to activate your LANCOM VoIP-Option again, for example after a repair.

Help in case of problems

If you have problems with registering your software option, please contact us by e-mail at optionsupport@lancom.de.

5.3 Activating the LANCOM VoIP-Option

Activating the LANCOM VoIP-Option is very simple. In LANconfig, mark the appropriate LANCOM (simply click on the entry with your mouse) and select the menu item **Device ▶ Activate software option**. Alternatively, click on the entry for the device with the right-hand mouse key and select **Activate software option** from the context menu. In the following window, enter the activation code that you received with your online registration. The device will then restart automatically.

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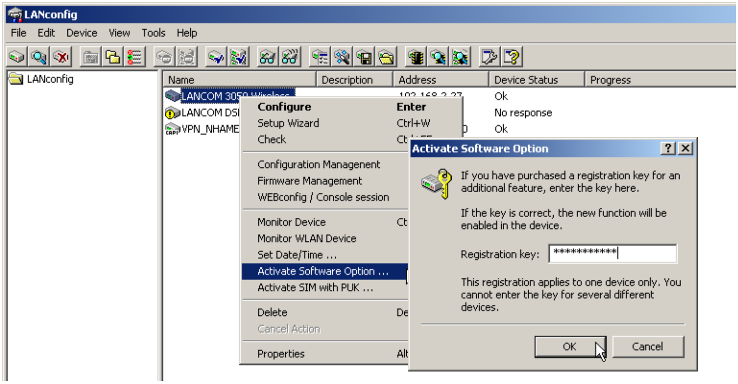
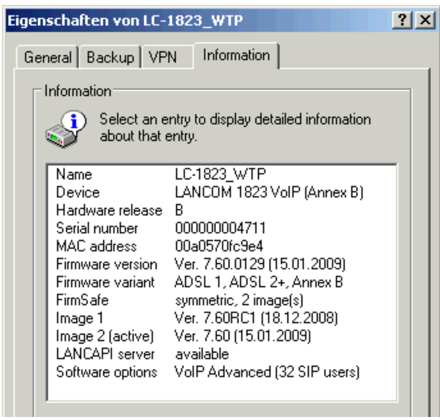


5.4 Checking the activation

You can check if the online activation of your LANCOM VoIP-Option was successful by selecting the device in LANconfig and selecting the menu item **Device** ▶ **Properties**. The properties windows contains a tab named 'Info' that lists the activated software options.



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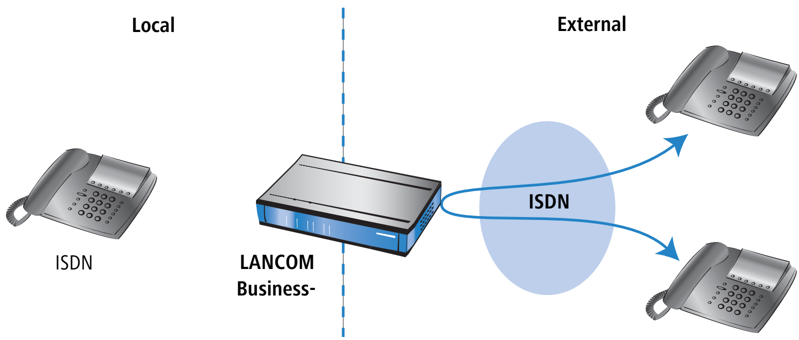
6 Extended functions

This section introduces extended functions that demand a more in-depth understanding of the PBX in the LANCOM Business-VoIP-Router and of PBXs in general. Please refer to the LCOS reference manual where necessary.

6.1 Setting up call forwarding in the telephone exchange

Within ISDN networks, terminal devices (telephones) use a set of control sequences—the so-called ISDN facilities—for communications with the exchange. These facilities allow, for example, call transfers to be set up at the telephone exchange. For example, if the exchange receives the sequence `*21*0123456789#`, all calls intended for the telephone number or MSN that issued the sequence will be forwarded to the telephone number "0123456789". Call forwarding at the exchange prevents ISDN channels from being blocked, unlike the case when call forwarding is handled by the telephone itself.

To provide a local telephone infrastructure with the functions of a PBX, the LANCOM Business-VoIP-Router must be able to process the ISDN-network control characters, and consequently these characters can no longer be forwarded to the exchange. The LANCOM Business-VoIP-Router itself becomes the exchange for any telephones connected to it and it manages all incoming and outgoing telephone calls. In case of call forwarding the sequence `*21*0123456789#` sent to the LANCOM Business-VoIP-Router instructs it to forward all incoming calls to the number "0123456789". The ISDN exchange does not have any information about this call forwarding.



The disadvantage of this variant is that call forwarding blocks two ISDN channels in the LANCOM Business-VoIP-Router, even though none of the local users is making a call.

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In order for call forwarding to be handled by the exchange in combination with the PBX in the LANCOM Business-VoIP-Router, three entries are required in the call-routing table:

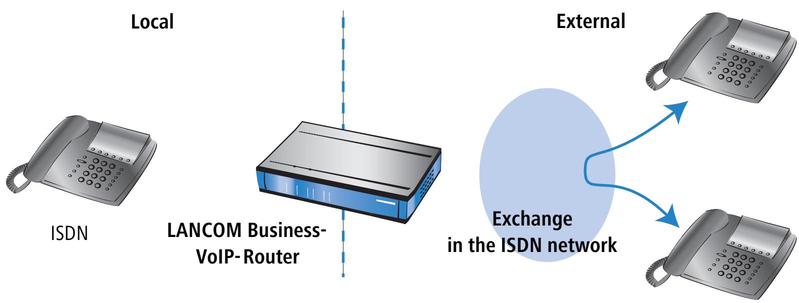
Called no.	Comment	Target no.	Destination line
890	CF switched off	#21#	ISDN
891	CF switched on	*21*0123456789#	ISDN
899	Check CF	*#21#	ISDN

- For the called number, enter a speed-dial number that is to transmit the ISDN facilities to the exchange, in this case '890' and upwards. The speed dial may not contain control characters (* or #).
- As target number you enter the ISDN facilities for communication with the exchange, e.g. `*21*0123456789#` to activate call forwarding.

Enter the external telephone number as a call-forwarding target just as you would dial the subscriber from the public telephone network. Even if you have deactivated spontaneous outside line access and you have to dial a zero for outside calls, the forwarding target here is entered **without** a leading zero.

- As the target line, enter 'ISDN' for each as this function only refers to the exchange in the ISDN network.

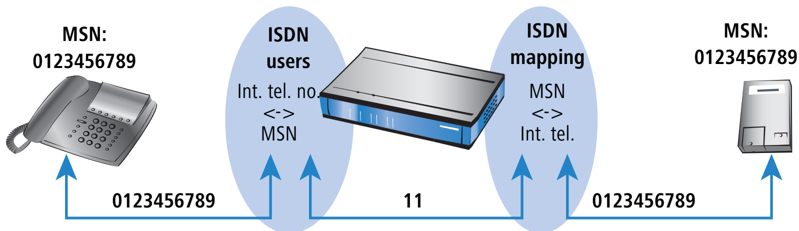
In this way, the speed dial for immediate, "unconditional call forwarding" can be complemented by other speed dials for "call forwarding on busy" and "delayed call forwarding" to trigger call forwarding at the exchange.



The procedure of call forwarding at the exchange is not visible at the LANCOM Business-VoIP-Router—consequently the forwarding of calls is not displayed in LANmonitor.

6.2 Life-line support for ISDN telephones

Life-line support allows LANCOM Business-VoIP-Routers to continue to provide telephony to any connected ISDN telephones, even if the LANCOM device is unconfigured and/or in case of power outage. To ensure that this function works, the telephone numbers are mapped twice between the ISDN NTBA and the terminal equipment.



1. At the transition from the ISDN NTBA to the LANCOM Business-VoIP-Router, the ISDN mapping table for the ISDN line defines the translation between external MSNs from the ISDN network and internal telephone numbers. For example, an incoming call for the MSN '0123456789' is converted into the internal telephone number '11'.
2. At the transition from the LANCOM Business-VoIP-Router to the ISDN telephone, the ISDN user table defines the translation between the internal telephone number and the internal MSN. The incoming call directed to '11' is converted back into '0123456789'. For this reason, the ISDN telephone needs this "MSN" programmed into it as its internal MSN in order for it to react to this telephone number.
3. If the LANCOM Business-VoIP-Router is not available as a telephone exchange (device is unconfigured or there is a power outage) then the the ISDN interfaces provide a "bridge" between the internal devices and the external connection.

A requirement for this function is the correct configuration of the ISDN interfaces and of the DIP switches on the underside of the device—these are the standard ex-factory settings.

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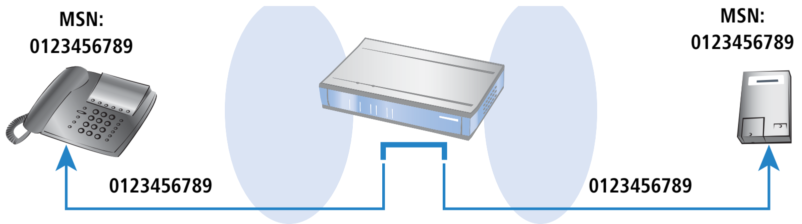
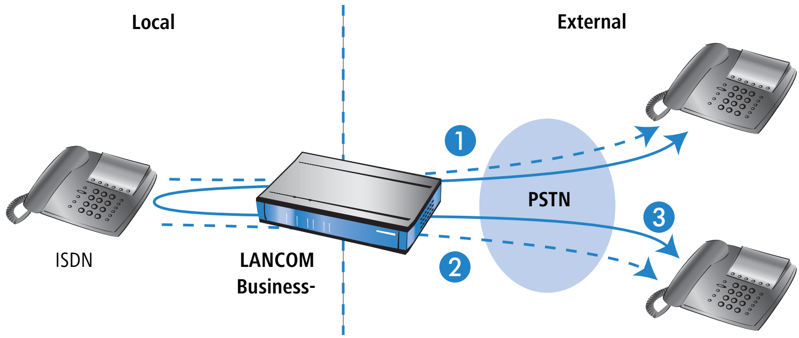
If the internal MSN in the ISDN user settings agree with the external MSN, then the ISDN telephone can be operated without the LANCOM Business-VoIP-Router just as if it were directly connected to the NTBA.



Further information on life-line support can be found in the user manual for your LANCOM Business-VoIP-Router and in the LCOS reference manual.

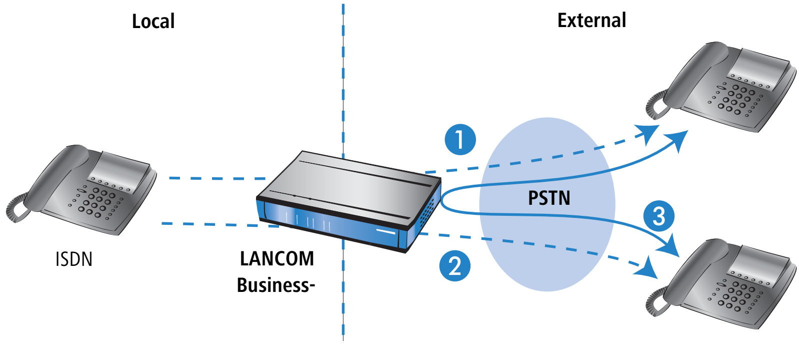
ECT setting

ECT (Explicit Call Transfer) is the spontaneous connection of two active calls by the user: If you are conducting two conversations, you can transfer the two callers so that they can then communicate with each other ('Spontaneous call management by the user' →Page 29).



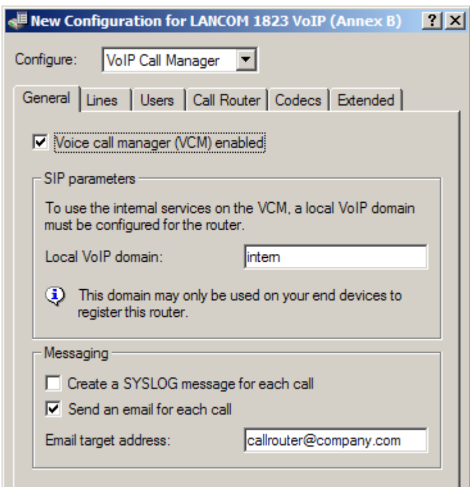
An ISDN telephone can transfer and interconnect the two active calls itself—even without a PBX. The ISDN telephone connects the two calls with each other internally and the two callers can speak with one another directly. The disadvantage of this variant is that the two lines which would normally be available on the ISDN telephone are both engaged. The user who connected the two callers can no longer conduct calls and is not able to receive calls for as long as the other two continue their call.

Alternatively, the ISDN telephone can delegate the transfer of the two active calls to the exchange. In this case the LANCOM Business-VoIP-Router works as the exchange, leaving the ISDN telephone lines free to conduct further calls.



6.3 Messages about calls

You can optionally receive information about all of the calls conducted via the LANCOM Business-VoIP-Router. In LANconfig you set up the messaging under **VoIP Call Manager ▶ General▶ Messaging**.



Here you define whether the messages are to be sent as e-mail and/or SYSLOG (Facility: Accounting; Level: Info). For every call which is connected (internal, external, incoming, outgoing), a message is generated containing information such as the source and target number, start-time and end-time of the call, etc.

For SYSLOG messaging set up a SYSLOG client (**LANconfig ▶ Log & Trace▶ SYSLOG**) and for e-mail messages set up an SMTP account (**LANconfig ▶ Log & Trace ▶ SMTP account**).

Please not that the information in these messages may be confidential in nature. In certain cases legal aspects of privacy, data privacy or labor law may apply.

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