Voice Server Anuncio for ISDN II & PRI

Manual

V2.1

VC2007

vidicode

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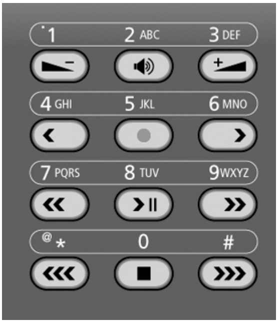
Recorder-keys desktop model

Record Backward Stop Pause/Play Forward Skip

Recorder keys 19” model



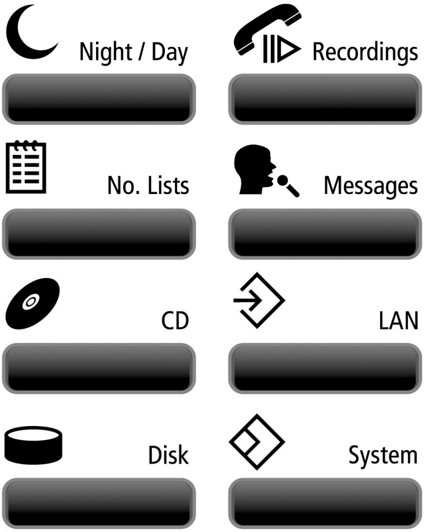
On the 19” model the numerical keys and the recorder keys are combined. This is possible because depending on the selected procedure the function of the keys is always obvious.

About the recorder keys on all models

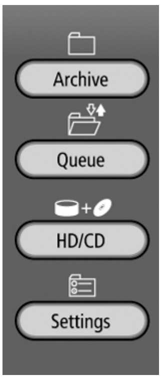
During recording and playback the recorder keys have the standard recorder key functions.

Otherwise the and keys can be used to scroll through a selection of recordings, numbers in a number list or through a settings menu. The key will always exit the current operation. When editing text they function as explained in the table of the alpha-numerical functions below.

Function-keys desktop model



Function keys 19” model

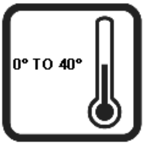


Alfa-numerical functions

Numerical keys	Function Press the relevant key repeatedly or hold down:				
	1x	2x	3x	4x	5x
. 1	1	.	:	,	;
ABC 2	A	B	C	2	!
DEF 3	D	E	F	3	%
GHI 4	G	H	I	4	^
JKL 5	J	K	L	5	$
MNO 6	M	N	O	6	@
PQRS 7	P	Q	R	S	7
TUV 8	T	U	V	8	&
WXYZ 9	W	X	Y	Z	9
@ *	*	@	/	<	>
0	0	_	+	-	=
#	#	(	)	[	]

Recorder keys	Function
»»	Toggle uppercase and lowercase
»	Forward
«	Backward

Care and Maintenance

	Keep the Voice Server dry. If it gets wet, wipe it dry immediately with a soft, clean cloth. Liquids might contain minerals that corrode the electronic circuits.
	Use and store the Voice Server only in temperature conditions between 0 and 40 degrees Celsius. Temperature extremes can shorten the life of electronic devices and distort or melt plastic parts.
	Keep the Voice Server away from excessive dust and dirt.
	Do not use aggressive chemicals, cleaning solvents or strong detergents to clean the Voice Server.

Contents

1 Introduction

The main features of the Anuncio are:
Call Queuing, Call Transfer and Call Statistics.

There are two PC programs available: One to configure the Voice Server Anuncio over the network and another for the Call Statistics, to analyse and produce displays as graphic or numeric charts.

The set up of the Voice Server can be done with the Voice server Configuration software or on the keyboard on the VS itself. For more complex settings however, it is recommended to use the special Voice server Configuration software. This software is connected with the Anuncio through the LAN and is easy to use.

The Anuncio is normally connected between the NT (network termination point) and the PBX. However, the Anuncio can also operate "stand alone", connected only to the NTP or behind a PBX on a suitable ISDN extension card.

Voice Server Anuncio is made for basic rate (BRI) or primary rate (PRI) ISDN lines. There are three types available as desktop (BRI only) or 19” models:

- Voice Server PRI for one E1 line for up to 30 channels
- Voice Server BRI for two S0 lines for up to 4 channels
- Voice Server BRI for four S0 lines for up to 8 channels

A Voice Server always has an internal drive and a network interface.

This manual applies to all variations of the Voice Server Anuncio. The operation is kept almost the same between models.

1.1 Basic functions

The Anuncio provides all common announcer services used in telephone systems:

1.1.1 Call Transfer and Call Queue

The Anuncio can transfer a call to an extension. This can be automated and influenced by the schedule or based on the selection in the call attendant menu. When a call is transferred this can be announced with a spoken message. When stations are busy the Voice Server can create a queue of callers waiting to be served. The caller will be informed about the progress that he makes in the queue. In between he can listen to music or spoken information.

1.1.2 Call Statistics

A Voice Server is usually bought to cope with an existing problem in the handling of telephone calls. Many problems however are unknown. Quite often it is unknown how many calls remain unanswered and how they are distributed over departments or times of the day. To come up with a good solution it is important that you analyse the data of a certain period, of departments, etc.

After installing the Anuncio it is important to keep track of the call statistics. When you create a call queue for example and the callers have to wait too long in the queue you will loose them.

1.2 Network

The Anuncio features an Ethernet interface. It can be connected to a PC or computer network through the Ethernet interface. With its own IP address it will act as an FTP-server from which files can be downloaded using any FTP client.

The network interface has many applications:
The Anuncio can be configured and controlled from a PC in the network. For many users this will be the preferred way to configure the Anuncio. Configuration software for PC's with Microsoft Windows is included with the product or can be downloaded from our website www.vidicode.com.

1.3 Pass Through connection system

A Voice Server will be connected between the public telephone network and your telephone system. All ISDN channels pass through the internal bus of the Anuncio and it can take control at any time. Pass Through is also occasionally called In-Line.
As a consequence of the pass through connection the Anuncio can also intercept the D-channel and therefore control signalling between the telephone system and the network. This is used to filter numbers sent by the PBX so that they are not sent to the line.
It can be used to capture commands from the user to the recorder which has several applications that will be explained in this manual.
Because of pass through mode the Anuncio is equipped with loop-through relays to prevent disrupting telephone communication in case of a power failure or another malfunction. The loop-through relays then

close to fall back into passive mode and ongoing telephone calls are not disconnected.

2 Installation

2.1 What is in the box?

The following parts are supplied with your Voice Server Anuncio:

19” model PRI	Mains cable (not available for Switzerland and Australia) ISDN cable with two RJ45 connectors Brackets with screws for rack mount This manual
Desktop model BRI	Mains power supply adapter and cable 2 or 4 ISDN cable with two RJ45 connectors (2 meter) 2 or 4 ISDN cable with two RJ45 connectors (30 centimetres) 2 or 4 ISDN T-adapters Headset This manual
19” model BRI	Mains cable (not available for Switzerland and Australia) 2 or 4 ISDN cable with two RJ45 connectors (2 meter) 2 or 4 ISDN cable with two RJ45 connectors (30 centimetres) 2 or 4 ISDN T-adapters Brackets with screws for rack mount Headset This manual

2.2 Connecting the power supply

2.2.1 On the desktop model

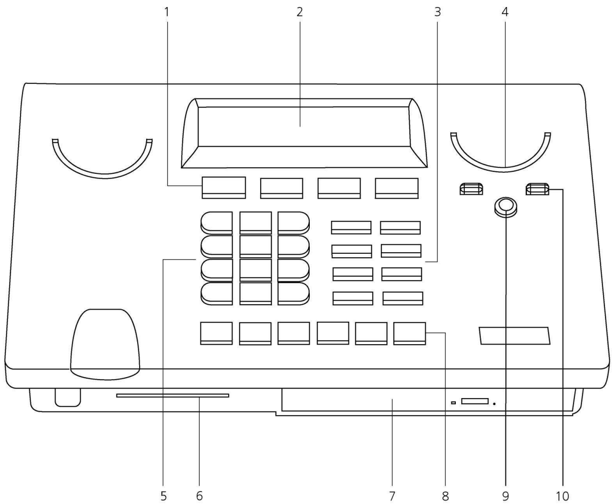
The Anuncio desktop model is only available for the BRI version and is powered through an AC power adapter. It does not have an On/Off switch; the Voice Server is turned on by connecting the power supply.

2.2.2 On the 19” model

Use the supplied mains cable. The Anuncio does not have an On/Off switch; it is turned on by connecting the power supply.
In some countries (Switzerland, Australia) no mains cable is supplied because it is not available in the country of manufacture (The Netherlands). Any standard computer mains cable will do.

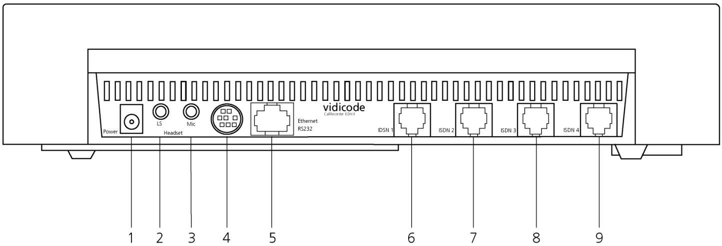
2.3 Connecting the ISDN lines

Voice Server Anuncio desktop model



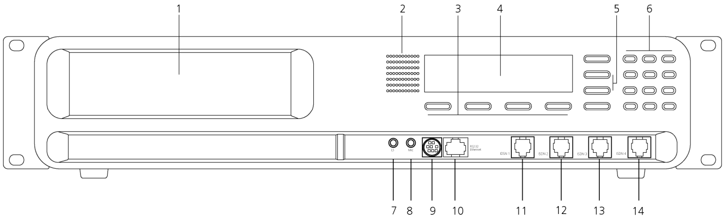
1. Menu keys
2. Display
3. Function keys
4. Speaker
5. Alfa-Numerical keys
6. CryptoCard reader (not used)
7. CD Drive (unavailable)
8. Recorder keys
9. Speakerphone key
10. Volume adjustment

Voice Server Anuncio Desktop model ISDN II Connections



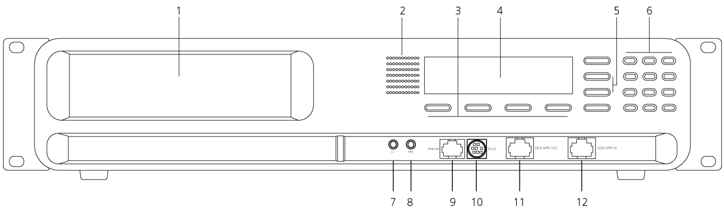
1. Power
2. Loudspeaker
3. Microphone
4. RS232 serial connection
5. Ethernet/LAN
6. ISDN BRI 4
7. ISDN BRI 3
8. ISDN BRI 2
9. ISDN BRI 1

Voice Server Anuncio ISDN II 19” model



1. CD-drive (unavailable)
2. Internal loudspeaker
3. Menu keys
4. Display
5. Function keys
6. Alfa-Numerical / Recorder keys
7. Loudspeaker
8. Microphone
9. RS232 serial connection
10. Ethernet/LAN
11. ISDN BRI 4
12. ISDN BRI 3
13. ISDN BRI 2
14. ISDN BRI 1

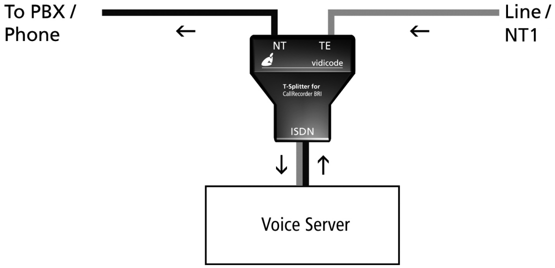
Voice Server Anuncio PRI 19”model



1. CD-drive (unavailable)
2. Internal loudspeaker
3. Menu keys
4. Display
5. Function keys
6. Alfa-Numerical / Recorder keys
7. Loudspeaker
8. Microphone
9. RS232 serial connection
10. Ethernet/LAN
11. ISDN PRI Out
12. ISDN PRI In

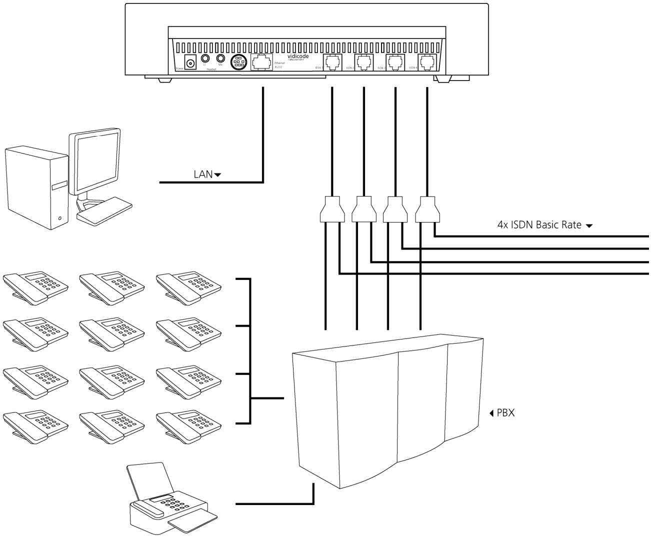
2.3.1 Connecting to Basic Rate ISDN

The Voice Server BRI is connected in series between the NT1 line port and other equipment

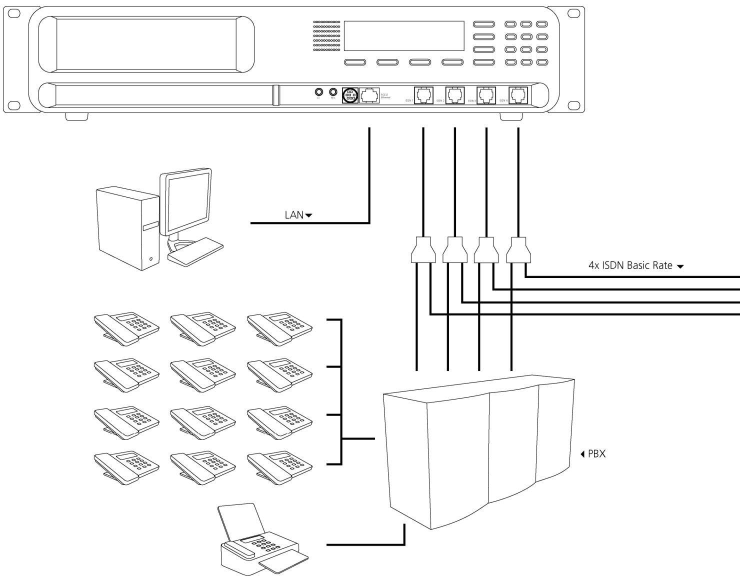


Per line you have received a T-splitter a long 8 wire ISDN cable and a short 8 wire ISDN cable.
Take the line of your existing equipment from the NT1 box and connect it to the NT port of the splitter.
Use one cable to connect the port labelled "ISDN" on the splitter with the port labelled ISDN 1 on the Voice Server.
Connect the port of the splitter labelled TE with the NT1 box.
Make the same connections for the ports ISDN 2, 3 and 4.
The figure is an example of a possible setup of the Voice Server BRI.

Connections ISDN II– desktop model



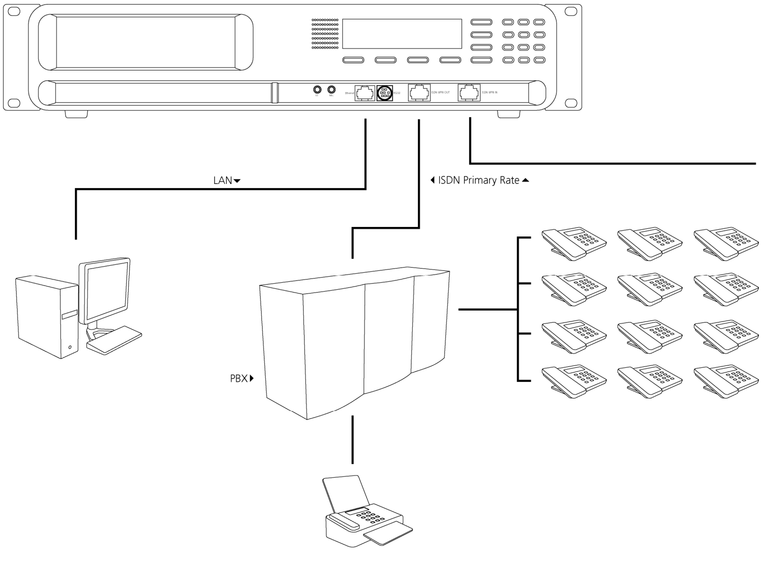
Connections ISDN II– 19” model



2.3.2 Connecting to E1 Primary Rate ISDN

The Voice Server PRI is connected in series to the E1 ISDN line between the line port and other equipment. With the Voice Server you have received a cable with two RJ45 connectors. This cable is used to connect the Voice Server PRI to the wall socket. The existing cable can be used to connect the Voice Server PRI to the PBX as is shown in the figure below.

Connections Anuncio PRI – 19” model



2.4 Connect the network

The network is connected using a generic network cable. This cable is not supplied with the product.

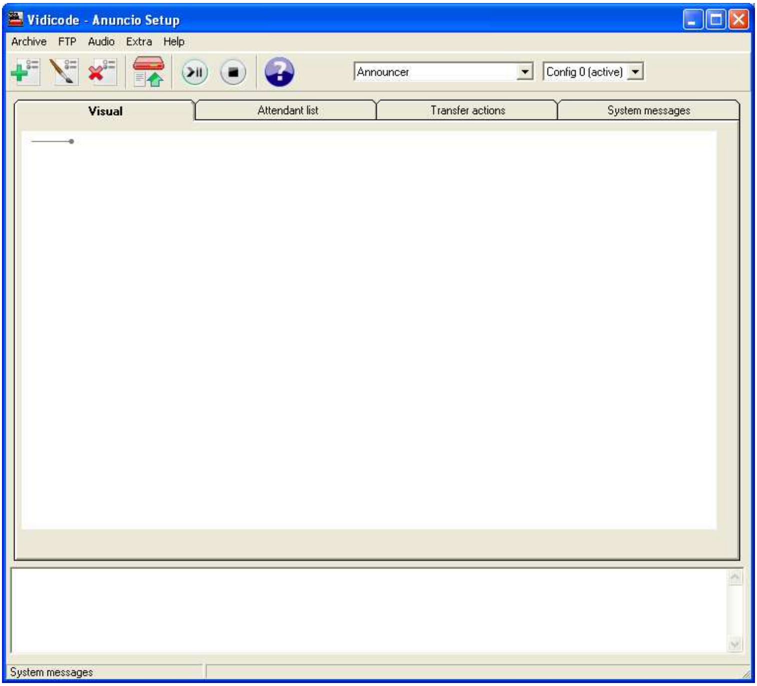
2.5 Disconnecting ISDN

In all of the situations pictured above, you are leading the telephone lines through the Voice Server. This means that you temporary disconnect the telephone system from the network. Therefore you may want to connect the voice server after office hours.
When you have connected the Voice Server you must check if the telephone can be used again for external calls. If you have followed the instructions this should not be a problem. The Voice Servers assumes the most common wiring pattern. Especially with a Primary Rate line the wiring might be different because wiring is not specified as part of the ISDN standard. If there is a problem, nothing will be damaged but you may need assistance from your dealer to provide the correct cable.

3 Voice server configuration software

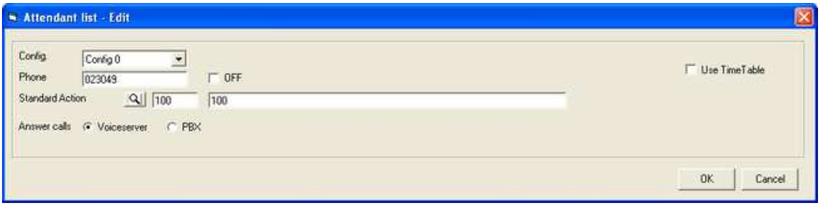
The easiest way to configure the Voice Server Anuncio is using the Voice server Configuration software. However, the Anuncio System setup has to be done first, on the Anuncio itself. Please read Chapters 4 and 5 and follow the instructions, after that, return to this chapter.

The software is on the CD and starts automatically when the CD is entered in the CD drive of the PC. The installation key is provided with the software.
This is what you will see after the Voice server Configuration software is installed.



3.1 Attendant List

Please read Chapter 7.1 for detailed information on the Attendant list. The Attendant List is a list of phone numbers that are handled by the Announcer, each entry in the Attendant List has 2 fields, a phone number and a action number. When the phone number is called, the Announcer takes the call and starts the message belonging to that phone number. Add an "N" at the front of the phone number, to specify an attendant for night service.



An action can optionally contain a spoken message (recording). First the message is played, and then the action is taken. If no spoken message (recording) exists then only the action is taken. Nothing is heard in that case. Possible actions are: Wait for selection , Connect Through or Record a voice-mail message. The action depends on the message type. If no action is defined, it is ignored and a connection to the called number is made.

An attendant belongs to one of the ten configurations.

Instead of a single action number a Time Table can also be connected to a phone number in the Attendant List. A timetable defines a whole week of actions. Each day of the week within that table can be filled with times and action numbers. The Call Attendant for that phone number can take different actions on different times and days.

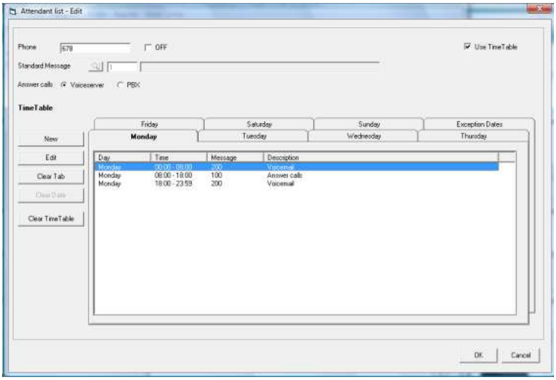
A phone number can also be set to action number "OFF". This means that no action is taken. This can be used to exclude phone numbers when using wildcards for others further in the list.

If you want to pass calls to a phone number to the PBX first, click on PBX in the radio group. This means that the attendant doesn't take the call immediately, but the call is passed to the PBX first. The attendant will "take back" the call if the called number is busy or doesn't answer for some time. Only in those two cases the action number behind the explanation mark is used and started.

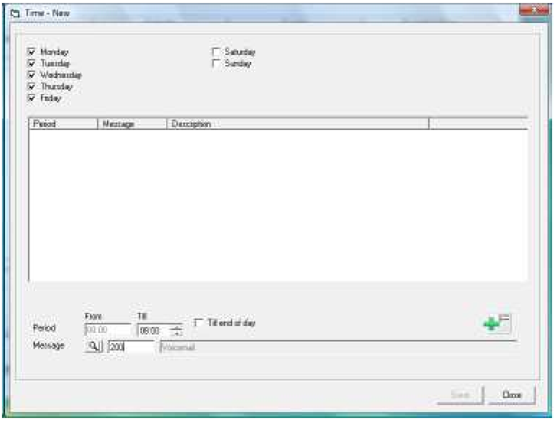
3.1.1 Time table

Time table files contain the answering schedule of an attendant number. You can create a repeating schedule for weekdays. With Exception dates you can create a schedule for a particular date or period

The timetable can be accessed through the attendant list by selecting Use Timetable

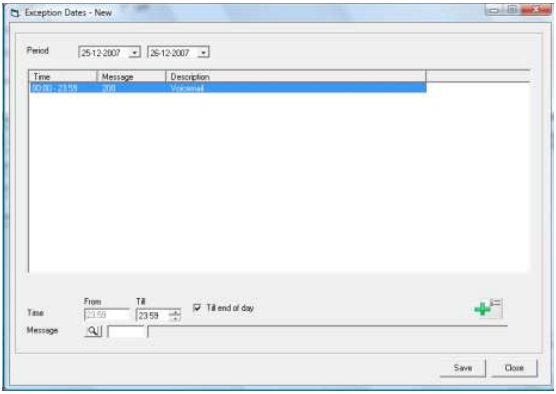


Creating a Time Table



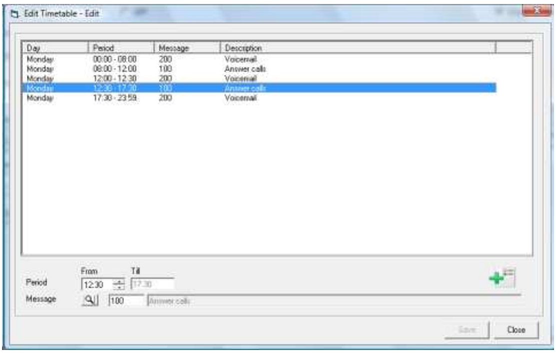
Step 1 - Select the day(s)
Step 2 - Configure the time period(s)
Step 3 - Save the data

Creating a Time Table for an exception date or period



Step 1 - Fill in the period
Step 2 - Configure the time period(s)
Step 3 - Save the data

Edit a Time Table



Step 1 - Select the period you want to edit
Step 2 - Change the start time of the period
Step 3 - Save the data

3.2 Transfer Actions

There are three types of transfer actions:

1. Direct Connect
The message is played and in the mean time the Announcer dials the number. The message stops as soon as the call is taken. If the message ends before the call is taken, the phone alerting/ringing tone is heard.

2. Notification
The message is played, and after it ends, the Announcer dials the number. The caller must listen to the whole message before he/she is connected.

3. Call Queue
The message is played and after that, the caller is placed in the call queue. There are two queue modes, depending on the "Phones in Queue" setting:

0 - Auto mode: The announcer dials the PBX every 3 seconds. The PBX must return 'busy' if all phones behind this number are occupied.

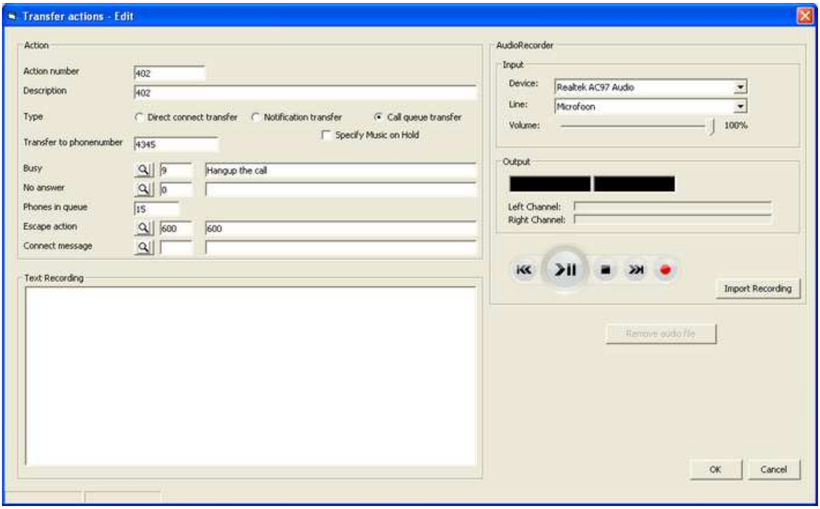
1-30 - Represents the number of available telephones (users) behind the queue number. The announcer itself keeps track of the busy lines and only dials the PBX after a phone is/becomes free.

During waiting the Announcer uses System Messages 050 till 071 and the On Hold Beeps. It is also possible to use music-on-hold. Create and Upload the Music on Hold System Message and in this form select the checkbox use music on hold. See also Chapter 8.2.1

More then one queue is possible: The original called phone number will distinguish between them. If no type is defined for a transfer action, it will act as Direct Connect. All transfer actions can change the phone number. That optional number is then used to dial the PBX. If no number is entered then the original called number from the incoming call is used.

For each transfer action a Busy and/or No Answer action number can be entered. If the dialed number is busy or doesn't answer for some time, the Announcer can take back the call and jump to another action number. The No Answer Timeout can be changed in the normal installation menu.

Note: A queue or busy-action can only work if the PBX returns the busy-status if that called phone is occupied.
Note1:
A queue or busy-action can only work if the PBX returns the busy-status on ISDN-level if that called phone is occupied.
Note2:
It is possible to define a call queue for multiple dial-in numbers (different called phone numbers).
The 'phone number' in the transfer action must start with "#" ("#1" defines queue 1). The original called number is not changed then when the PBX is dialed.



3.3 System messages

System messages are predefined messages such as beeps and call queue notifications used by the Announcer. You can record or import new audio for these system messages. See also Chapter 9

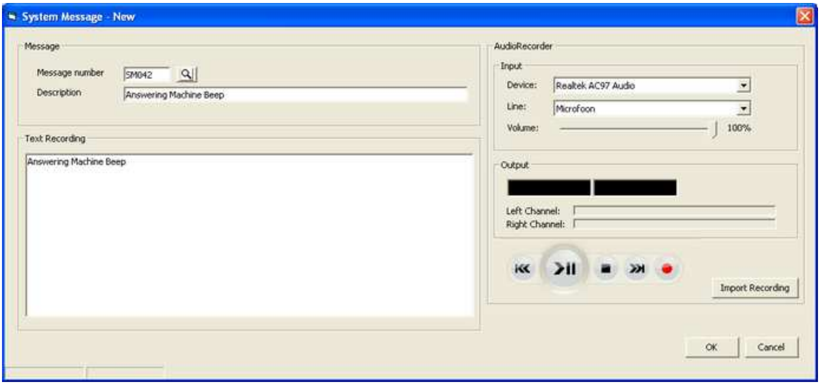
The following System Messages are in use:
039 = "One moment please..." (used after an input when waiting for a response from the network)
040 = Special Notification Message that can be included in recorded calls.
042 = Answering Machine Beep
043 = On Hold Beeps used in the Call Queue
044 = Phone Alerting Tone

045 = Enable/Action Beeps used for various functions (high tones)
046 = Disable Beeps (low tone)

System Messages 50 till 71 are optional and used for the Call Queue:
050 = "You are the first person to be transferred"
051 = "There is 1 person waiting before you"
052 (till 070) = "There are 2 (till 20) persons waiting before you"
071 = "There are more then 20 persons waiting"

If the Call Queue messages are not available then the On Hold Beeps are used instead.

System Messages 80 till 99 are also optional and used only in a Call Queue, replacing 50 till 71:
080 till 089 = Music-on-hold or advert files that are played sequentially, always starting with 080.
At least 080 and 081 must exist for this to work properly.
090 till 099 = Music-on-hold sound files.
If not existing then the system looks for file MUSIC.WAV.
If that doesn't exits then the normal on-hold-beeps are used.



Import Recording
Use this option to import a recording from an existing .WAV file.

4 Getting started on configuration

After you have connected the hardware it is time to configure and use the Voice Server. This chapter explains how the user interface works. In the following chapters we guide you with the configuration. You must:

- Go through the Voice Server settings and make changes if required
- Go through the LAN configuration and configure the Voice Server to work with your network

When the Voice Server is configured, you can set up the procedures of your Voice Server. This can be done from the keyboard of the Voice Server or via PC software. Please read Chapter 3.

- Add the numbers that should trigger the voice server to the "attendant list"
- Make procedures for these numbers or groups of numbers
- Go through the network settings so the Voice Server will fit in with your network and connect to the mail server

A "procedure" is nothing more then a series of spoken messages with their effects. These effects can be a next spoken message, transfer of the call, queuing of the call etc.

4.1 Operation basics

The Voice Server has a menu based user interface. Most functions and operations are initiated by pressing a function key followed by a sequence of menu keys of which the function is determined by the text in the display. The use of function keys and menu keys is as follows:

4.1.1 Function keys

The frequently used functions have been grouped in the function keys. Symbols representing the function keys are used to indicate the key corresponding to the described function.

4.1.2 Soft keys

After pressing a function key the available functions are assigned to the four menu keys. The operation of the menu keys is determined by the text in the display right above the corresponding key. Throughout the manual figures representing the display are used to show the required action corresponding to the described feature.

4.1.3 Frequently used key indicators

The following soft key functions are consistently used throughout the manual.

MENU in the display indicates the presence of an underlying menu

- Press **MENU** to open the underlying menu

NEXT in the display indicates the presence of more menu items

- Press **NEXT** to jump to the following menu item

BACK in the display indicates the presence of an embracing menu

- Press **BACK** to return to the embracing menu

STOP in the display indicates the absence of an embracing menu

- Press **STOP** to return to operating mode

CHANGE will toggle the item in the display between "On" and "Off"
In some cases CHANGE is used to increase or decrease a value.

4.1.4 Save Changes

After making settings the user is forced to press YES or NO before returning to operating mode. Press STOP in the menu and the display will show:

```
          Settings
       Save changes?

 YES      NO            CANCEL
```

YES returns the Anuncio to normal use with changed settings
NO returns the Anuncio to normal use without changed settings
CANCEL returns to the settings

4.2 ISDN procedures

The guiding principle of the Voice Server is that for the PBX it behaves as if the facilities are available in the ISDN telephone network.

The Voice Server can only control the PBX as if it were the network. This means that the Voice Server should send MSN numbers (Multiple Subscriber Numbers) to the PBX to make connections.

For various reasons it is possible that you run out of MSN numbers. Your PBX will only answer calls that it was told to answer. Normally the PBX will only answer calls with MSN numbers you received from your network provider. When you have a Voice Server you do not have to pay the provider for extra MSN numbers you want to use with the Voice Server. If you need extra MSN numbers just “invent” them. These virtual MSN numbers can not be dialled from the outside.

For the Voice Server to work, there must be an MSN number programmed in the PBX for every extension that has to be serviced by the Voice Server. These MSN numbers can be either real or virtual.

On the incoming side, the Voice Server can only respond to the number it has received. The Voice Server has a so called Attendant List where it finds a procedure for the number that has been called.

To summarise

- A telephone number for incoming calls has to be one of the MSN numbers given to you by the provider.
- If a number has to be **sent to the PBX**, it must be an MSN number as well. However, it does not have to be an MSN number given to you by the provider. This number should also be added to the list of MSN numbers that your PBX “knows”.

Some network providers give their users more flexibility. With these networks it is possible to extend the number dialled after the connection is made. The Voice Server can work with this procedure as well and will treat the received number together with its extension as the MSN number.

When configuring the Anuncio the subscriber number will be sufficient, because the Voice Server only used the last 5 digits.

5 System configuration

The system settings determine the basic functions and operation of your Voice Server. Fill them in guided by this chapter.

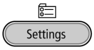
5.1 Opening the System menu

On the desktop model:

- Press the 

key to enter the recorder settings menu.

On the 19” model:

Press the 

key

- Press the soft key **SYSTEM** to move to the recorder settings menu

5.2 Call Statistics

The meaning of Call Statistics is that the Anuncio will make a record of every call, even unanswered calls. The data can be analyzed using the Call Recorder Access System. This is important because call statistics provide you with detailed knowledge about how efficient your company is in answering the telephone.

```
Call Statistics:  On
NEXT               CHANGE   STOP
```

5.3 Answering time

This is the time between the arrival of the call and the Voice Server answering (when it is instructed to answer immediately). A very short time gives a quick response, but to the caller it might be more common to hear the phone ringing at least once.

```
Answering Time:  0.2s
NEXT     - CHANGE +     STOP
```

- Press **+** or **-** to increase or decrease the Answering Time by 0.1 second.

5.4 No Answer Timeout

This is used when an extension is not answered. The no answer timeout is the time between the arrival of the call and that Voice Server answering. The Voice Server will only answer after the No Answer Timeout if it is instructed to do so.

```
No Answer Timeout:  16s
NEXT     - CHANGE +     STOP
```

- Press **+** or **-** to increase or decrease the No Answer Timeout by 1 second.

5.5 Connection

This setting only applies to an Anuncio for Basic Rate ISDN. It will not appear when your line type is Primary Rate.
There are two different line configurations possible for Basic Rate ISDN; point-to-point and multipoint. Point-to-point is common when several lines are connected to one PBX. Multipoint is for home users and small businesses when several machines are connected to one line (fax, PC, telephones). The signalling on the line is quite different and therefore the Anuncio must be configured for either one or the other. When you have a PBX it could still be possible the lines are multipoint because most PBX's can work with both systems. When in doubt, ask your supplier.

```
Connection:  Point-to-point
NEXT               CHANGE   STOP
```

5.6 Total lines

This setting only applies to an Anuncio Basic Rate. It will not appear when your line type is Primary Rate.
Configure the number of lines the Anuncio acts on. This can be 2, 4, 6 or 8. If you use port 1 only, it must be set to 2, if you use port 1 and to 2, it must be set to 4, etc, up to 8.

```
Total Lines:  8

NEXT      - CHANGE +     STOP
```

- Press **+** or **-** to increase or decrease Total Lines by 2.

5.7 Protect the Voice Server with a Password

The Voice Server can be protected with a Password against unauthorized use. When password protection has been enabled all actions that change the operation are blocked before the correct password has been entered.

```
Password: Off

NEXT                CHANGE   STOP
```

- Press **CHANGE** to enter a password.

```
New Password: ▒000

STORE                       CANCEL
```

- Press **STORE**.

```
Repeat Password: ▒

STORE                       CANCEL
```

- Press **STORE** to enable the entered password.
- Press **NEXT** to continue in the menu with the Clock setting (§ 5.9)

If the Password has been set you will be prompted for the password before entering the menus and before playback of recordings.

5.8 Remove Password protection

```
Password: On
NEXT                CHANGE   STOP
```

- Press **CHANGE** to disable the password.

The password protection has now been disabled.

- Press **NEXT** to continue in the menu with the **Clock** setting.

5.9 Set the Clock

To set the **Clock** do the following:

```
Clock: Mon 22-04-14 11:03
NEXT                CHANGE   STOP
```

- Press **CHANGE** to change the setting of the clock.

```
Clock: ▒on 22-04-14 11:03
STORE                      CANCEL
```

- Press key's 1-7 to set the day of the week starting with 1 for Sunday.

The default date and time format is DD-MM-YY and MM:HH (when American has been set as language the format is changed to MM-DD-YY). Use the arrowed keys to move the blinking character left or right.

- Press **STORE** to store the changes or press **CANCEL** to return to previous values.
- Press **NEXT** to continue in the menu with the **Language** setting or press **STOP** to exit the **configuration** menu.

5.10 Set the Language

To set the **Language**:

```
Language:        English
                 CHANGE    STOP
```

- Press **CHANGE** until you find the preferred language.

There is only a small difference between the language settings English and American. Normally the date format is displayed according to the European style convention (dd-mm-yy). When American is selected this will become (mm-dd-yy).

6 Configuration of the Ethernet interface

The Voice Server Anuncio has an Ethernet port. The network interface supports the following protocols:

FTP server
- SMTP client for sending e-mail messages to a SMTP server
- Telnet for remote configuration
- NTP for automated adjustments of the system clock
- A propriety protocol for streaming audio for real time remote monitoring

FTP server is used by the Voice Server Setup program and the Call Recorder Access System. The FTP server can also be used to give other (custom) applications access to the recordings (on the FTP server) in the Voice Server. It is even possible to open the Voice Server as a network drive from a PC.
SMTP gives the recorder the ability to send e-mails. This has two applications. The recorder can send status updates to the systems manager.
Telnet is a possibility to give a systems manager access to internal menu's from remote.
The NTP (Network Time Protocol) is a good method to provide accurate time information.
Real time remote monitoring (RTRM) is available for users of the RTRM PC software. It will be described later in this manual.

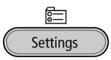
6.1 Network active

To begin the network configuration on the desktop model:

- Press the 

function key to enter the network settings.

On the 19" model:

- Press 

- Press LAN

The display will show:

```
Network active:   No
NEXT              CHANGE  STOP
```

- Press **CHANGE** to enable or disable the network.
- Press **NEXT** to move on to the next menu item.

6.2 FTP active

FTP stands for File Transfer Protocol.

```
FTP active:       No
NEXT              CHANGE  STOP
```

- Press **CHANGE** to enable FTP.
- Press **NEXT** to move on to the next menu item.

6.3 FTP user

The FTP user is the user name to be used by FTP clients such as the Call Recorder Access software to log on to the Voice Server.

```
FTP user:  0000
NEXT              CHANGE  STOP
```

- Press **CHANGE** to change the FTP user name.

```
FTP user:  ░
                        CANCEL
```

Use the numerical keys to enter the FTP user name. Use to switch between upper case and lower case characters.

- Press **STORE** to save the FTP user name.
- Press **NEXT** to move on to the next menu item.

6.4 FTP password

The FTP password is the password that goes with the FTP user name.

```
FTP pwd:  0000

NEXT              CHANGE   STOP
```

- Press **CHANGE** to change the FTP password.

```
FTP pwd:  ▒

                       CANCEL
```

Use the numerical keys to enter the FTP password.

- Press **STORE** to save the FTP password.
- Press **NEXT** to move on to the next menu item.

6.5 FTP server port

FTP server port is the port number through which an FTP client can log on to the Voice Server. The FTP server port is default set to 21, as is most common. If there is no direct reason to change the FTP server port it is best left unchanged.

```
FTP server port:  21

NEXT              CHANGE   STOP
```

- Press **CHANGE** to change the FTP server port number.

```
FTP server port:  ▒

                       CANCEL
```

Use the numerical keys to enter the FTP server port number.

- Press **STORE** to save the FTP server port number.
- Press **NEXT** to move on to the next menu item.

6.6 DHCP server

In case a DHCP server is used on the network the DHCP server option must be enabled. In case a DHCP server is not used on the network it must be disabled.

```
DHCP server:      Yes

NEXT              CHANGE   STOP
```

- Press **CHANGE** to enable DHCP server.

```
DHCP server:       No

NEXT              CHANGE   STOP
```

- Press **NEXT** to move on to the next menu item.

When a DHCP server is used, the IP and Gateway addresses are automatically assigned. Without DHCP server you must manually enter these IP addresses.

6.7 IP address

As part of the network the Voice Server needs an IP address. When a DHCP server is used (see previous item) the DHCP server will assign an IP address. When a DHCP server is not used a static IP address must be assigned to the Voice Server.

```
IP addr:  0.0.0.0

NEXT              CHANGE   STOP
```

- Press **CHANGE** to enter the IP address

```
IP addr:  ▒00.000.000.000

STORE                    CANCEL
```

Use the numerical keys to enter the IP address.

- Press **STORE** to save the IP address.

- Press **NEXT** to move on to the next menu item.

6.8 IP subnet mask

The IP subnet mask is used if access from outside the network is required. In this case the Gateway must be entered as well.

```
IP mask:  255.255.255.000
NEXT                CHANGE   STOP
```

- Press **CHANGE** to change the IP subnet mask

```
IP mask:  ▒55.255.255.000
STORE                      CANCEL
```

Use the numerical keys to enter the IP subnet mask.

- Press **STORE** to save the IP subnet mask.
- Press **NEXT** to move on to the next menu item.

6.9 Gateway

The Gateway is used if access from outside the network is required. If so the Gateway and the IP subnet mask must be entered.

```
Gateway:  0.0.0.0
NEXT                CHANGE   STOP
```

- Press **CHANGE** to enter the Gateway address.

```
Gateway:  ▒00.000.000.000
STORE                      CANCEL
```

Use the numerical keys to enter the Gateway.

- Press **STORE** to save the Gateway.
- Press **NEXT** to move on to the next menu item.

6.10 IP name

Aside from the IP address the Voice Server can also be addressed by an IP name if your DNS server supports this function.

```
IP Name:  BRI-FFFFFF

NEXT                 CHANGE   STOP
```

- Press **CHANGE** to enter an IP name.

```
IP name:  ▒

STORE                       CANCEL
```

Use the numerical keys to enter an IP name.

- Press **STORE** to save an IP name.
- Press **NEXT** to move on to the next menu item.

6.11 E-mail

Status reports and malfunctions can be e-mailed to this E-mail address.

```
E-mail:  john@vididcode.com

NEXT                 CHANGE   STOP
```

- Press **CHANGE** to enter an E-mail address.

```
E-mail:  ▒

STORE                       CANCEL
```

Use the numerical keys to enter the E-mail address.

- Press **STORE** to save the E-mail address.
- Press **NEXT** to move on to the next menu item.

In this manual different e-mail lists will be introduced. For the e-mail lists to be operable the main e-mail address has to be a valid -e-mail address.

6.12 Reply address

Because the Voice Server cannot receive E-mail, the E-mails sent require a reply address.

```
Reply:  john@vididcode.com

NEXT                CHANGE   STOP
```

- Press **CHANGE** to enter a reply address.

```
Reply:  ▒

STORE                      CANCEL
```

Use the numerical keys to enter the E-mail reply address.

- Press **STORE** to save the E-mail reply address.
- Press **NEXT** to move on to the next menu item.

6.13 SMTP server IP address

The Voice Server requires the IP address of the SMTP server to send E-mail. In case a local SMTP server is used the IP number can be entered directly. In case of an SMTP server outside the network both the **Gateway** and the **Subnet Mask** has to be set later on.

```
SMTP serv:  0.0.0.0

NEXT                CHANGE   STOP
```

- Press **CHANGE** to enter the IP address of the SMTP server.

```
SMTP serv:  ▒00.000.000.000
STORE                 CANCEL
```

Use the numerical keys to enter the IP address of the SMTP server. Either an IP number or IP name are allowed. In case an IP name is used the DNS server must be configured.

- Press **STORE** to save the IP address of the SMTP server.
- Press **NEXT** to move on to the next menu item.

6.14 SMTP-server port

SMTP-server port is the port number through which the connection is to be made with the SMTP server. The SMTP-server port is default set to 25. Do not change the SMTP-server port number when this value is not explicitly changed in the SMTP server.

```
SMTP serv port:   13
NEXT            CHANGE   STOP
```

- Press **CHANGE** to change the SMTP-server port number.

```
SMTP serv port:   ▒
                      CANCEL
```

Use the numerical keys to enter the SMTP-server port number.

- Press **STORE** to save the SMTP-server port number.
- Press **NEXT** to move on to the next menu item.

6.15 Domain name

When the SMTP server is outside the network the domain name of this server is required. Contact your provider for more information.

```
Domain:

NEXT                CHANGE   STOP
```

- Press **CHANGE** to enter the domain of the SMTP server.

```
Domain:  ▒

STORE                      CANCEL
```

Use the numerical keys to enter the domain of the SMTP server.

- Press **STORE** to save the domain of the SMTP server.
- Press **NEXT** to move on to the next menu item.

6.16 DNS server

When an IP name has been configured for your SMTP server you need to configure the IP number of the Domain Name Server.

```
DNS serv:  0.0.0.0

NEXT                CHANGE   STOP
```

- Press **CHANGE** to enter the DNS server.

```
DNS serv:  ▒00.000.000.000

STORE                      CANCEL
```

Use the numerical keys to enter the DNS server.

- Press **STORE** to save the DNS server.
- Press **NEXT** to move on to the next menu item.

6.17 NTP server

If there is a possibility on your network to give the Voice Server access to a NTP (Network Time Protocol) server, it is recommended you enable it because it will give the recorder an accurate time reference.

```
NTP serv:  0.0.0.0
NEXT               CHANGE  STOP
```

- Press **CHANGE** to enter the NTP server.

```
NTP serv:  ▒00.000.000.000
STORE                   CANCEL
```

Use the numerical keys to enter the NTP server.

- Press **STORE** to save the NTP server.
- Press **NEXT** to move on to the next menu item.

6.18 NTP port

NTP port is the port number through which the recorder as a client can connect to the NTP server. The NTP port is default set to 123. Consult the network manager for the port number.

```
NTP port:       123
NEXT               CHANGE  STOP
```

- Press **CHANGE** to change the NTP server port number.

```
NTP port:       ▒
                        CANCEL
```

Use the numerical keys to enter the NTP port number.

- Press **STORE** to save the FTP server port number.

- Press **NEXT** to move on to the next menu item.

6.19 GMT correction

GMT correction property is used to identify the time zone. NTP server normally issues GMT (also know as UTC). GMT correction can be set in half hours from -15:00 to + 15:00.

```
GMT correction:  00:00
NEXT      - CHANGE +    STOP
```

- Press **CHANGE** to set GMT correction. Keep pressing **CHANGE** to increase the correction. It will start with + 1:00, keep pressing **CHANGE** to increase. After + 15:00 – 15:00 will appear.
- Press **NEXT** to move on to the next menu item.

GMT correction refers to winter time. Summer time and winter time are corrected automatically.

6.20 Telnet

Telnet can be used to log on to the Voice Server and configure it from your PC. If required contact your supplier for more details.

```
TelNet active:    No
NEXT              CHANGE   STOP
```

- Press **CHANGE** to enable TelNet.

```
TelNet active:   Yes
NEXT              CHANGE   STOP
```

- Press **NEXT** to move on to the next menu item.

6.21 Service timer

The Service timer determines the performance of the network connection. Default the Service timer is set to Automatic. Changing the setting should only be done when advised by a service engineer to solve problems.

```
Server timer:    Auto

NEXT                CHANGE   STOP
```

- Press **CHANGE** to change the Service timer.

The following settings are available:
S1-S5, F1-F4 and U1-U4. Where S stands for Slow, F for Fast and U for Ultra fast.

- Press **NEXT** to move on to the next menu item.

6.22 Monitor active

The monitor function allows you to listen to the conversations that take place. This requires the RTR Call Monitor software, which connects with the Voice Server over the network.

```
Monitor active:   No

                    CHANGE   STOP
```

- Press **CHANGE** to enable the Monitor.
- Press **NEXT** to move on to the setting of the Monitor password.

The monitor password is used to log on to the Voice Server. This password must also be set in the RTR Call Monitor software.

```
Monitor pwd:    0000

                    CHANGE   STOP
```

- Press **CHANGE** to enter a password
- Press **STOP** to exit the network configuration.

7 The Attendant list

Press the Number list function key on a desktop model.
On a 19'' model press function key Settings and then soft key LIST.

```
        Attendantlist – 1/3
          123456 > 401

 NEXT     NEW     CHANGE     STOP
```

The example above is that there are three entries in the attendant list. The number 123456 is connected to message 401. So when someone calls the number 123456 he will be answered by message 401.

Eventually the attendant list will contain all numbers that you want to be answered by the call attendant of the Voice Server.

With **NEXT** you can browse through the list. With **CHANGE** you can change the selected number. With **NEW** you can add a new number to the list.
When you select **NEW** you see the following:

```
 Phone number:

          NIGHT            CANCEL
```

Day and night service may be different entries in the Attendant list. When you press **NIGHT**, an **N** will appear in front of the number.

For every entry in the Attendant list is the possibility to create a schedule with instructions, weekdays and times of the day. This is yet another option for the same telephone number

It is important to understand the difference between day and night service and the schedule. The schedule is automated, but therefore fixed. Day and night service must be set manually, every day.

It is possible to have different schedules for day service and night service. That means, that there are 4 possibilities:

- DAY service

- NIGHT service
- DAY Schedule
- NIGHT Schedule

When the Voice Sever is in DAY service only DAY entries from the Attendant list will be used. A schedule has priority over normal entry. To give an example: Suppose you want to answer with page 100, but at lunchtime you want to answer with page 200. Then you must refer to 100 for DAY service. Then you make a schedule that contains an entry 13:00 = 200 and 13:45 = ? for every workday.

- When you want to enter the schedule, you press **CHANGE**:

```
      Attendantlist - 1/3
          123456 > 400

 TABLE  NUMBER   DELETE   BACK
```

- Press **BACK** and **NEW** and create a new entry for 123456.
- Press **TABLE**.

```
   Timetable for 123456
 Monday    00:00=?

 DAY     >>     CLEAR     BACK
```

Following the example above you enter

```
   Timetable for 123456
 Monday    13:00=200

 DAY     >>     CLEAR     BACK
```

And

```
   Timetable for 123456
 Monday    13:45=?

 DAY     >>     CLEAR     BACK
```

Pressing **DAY** repeatedly will rotate the days of the week and the exception dates. A new exception date can be recognised by:

```
   Timetable for 123456
??-??          00:00=?

DAY     >>     CLEAR    BACK
```

If you want a different message on New Year's day for example:

```
   Timetable for 123456
01-01          00:00=301

DAY     >>     CLEAR    BACK
```

Upon answering the Voice Server will:

1. See if **DAY** or **NIGHT** service is selected,
2. when **DAY** is selected, first look at the timetable for **DAY** service for the number that was called.
3. If there are no instructions in the timetable follow the standard instruction

7.1 More about the Attendant list

The Attendant list determines how and when the Anuncio answers incoming calls. The Anuncio can hold up to 10 different lists, each containing a pre-defined setup. The user can select an active list by pressing 0 to 9 in the Attendant list menu on the Anuncio itself or with a DTMF code from a local telephone or with the special PC software.

Each entry in the Attendant list has 2 numbers: A telephone (extension) number and a message number or timetable.

The telephone number must be able to identify a local DDI number. When the number is called, the Anuncio starts the message number assigned to that telephone number. The message number can be changed by day and time using a timetable.

The message number can include an audio file but must at least have an action to be taken. First Playback of the Audio file starts, then the action is taken. In some cases the action is "held" until the playback of the audio file has finished

Possible actions are: Wait for a selection, Send the call to the PBX (connected between the PBX and network) and Record a message.

If no action is defined for a message number the call is sent to the PBX using the called number. For consistency there should always be an action even if the default action is the one desired.

A timetable (or schedule) defines a whole week of actions. Each day of the week within that table can be filled with times and message numbers. It can also contain exception dates, such as public holidays (New Year, Christmas, etc).

If no message number is entered for a certain time then the system looks in the Attendant list for a second entry. The same phone number can occur twice in the list. The order is important; the first entry must be the timetable and the second must be with a message number.

A telephone number can be set to the message number "Off" meaning no action is to be taken. This can be used to exclude phone numbers when using wildcards below it in the list.

Special case 1:
Pressing * when entering the message number displays an exclamation mark (!) in front of the number. In this case the attendant doesn't process the call immediately and the call is passed directly to the PBX. The attendant will "take back" the call if the called number is busy or doesn't answer for some time. When the call is “taken back” the message number behind the exclamation mark is used and started. The same effect can be achieved using a direct connect transfer (see below) without an audio file. Using this method the attendant first interacts on an isdn-level.

Special case 2:
The telephone number "00000" (5 zeroes) invokes special processing for numbers in the Attend list when there is no calling party number (no CLI or the number has been withheld) for a call. The message number specified is used instead of the associated message number.

Note:
The length of the numbers in a list must be the same. A wildcard suffix can be used (numbers with *) to provide the means to enter a common part and then any trailing digit is accepted without entering each and every instance. The lists are processed from top to bottom allowing exceptions to wildcards to be placed higher in the list. For a called

number the matching process starts from the right digit and proceeds leftwards.

The numbers in the Attendant list must be at least 2 digits long. The numbers length need only be that which makes a called number unique. For example, the DDI numbers purchased may consist of some, or all, the numbers from 0793471000 to 0793471099. In the example it is not needed to enter the complete telephone number each time but only the last 4, 5, 6 or 7 digits.

In some countries the number is made up of an area code and then subscriber number inside that area. The called number will then only be the subscriber number. Including any part of the area code in the numbers entered in the Attendant list will prevent matching.

8 Transfer messages

Like Attendant menus, Transfer messages are a spoken message with instructions to the Voice Server. As with a menu you must add instructions to the message. There are three types of Transfer messages:

Direct connect
The message is played while the Voice Server dials the number. The message stops as soon as the call is answered. If the message ends before the call is taken, the phone is heard ringing.

Notification
The message is played and only after it ends the Voice Server dials the number. The advantage is that the caller always hears the complete message. This can be important when a message is used as a recording notification.

Call Queue
The message is played and then caller is placed in a queue. While waiting in the queue he hears progress notifications, in between the progress notifications he can listen to "music on hold".

With the Transfer message types **Direct Connect** and **Notification** it is possible to define what happens if the number is busy or does not answer. The Anuncio can take the call back and jump to another message number.

8.1 Direct Connect and Notification messages

These Transfer messages are rather similar.

- Press the **Message** key and you will see:

```
  Announcements & Transfers
     (press PLAY or REC)

NEXT   [400]    CHANGE   STOP
```

- Type a number from 400 to 699 e.g. 425. You will see

```
    Announcements & Transfers
       (press PLAY or REC)

NEXT   [425]    CHANGE   STOP
```

- Record the message as you are used to do using the recorder keys, e.g.:

  "You will be connected to the support desk."

- Press **CHANGE**

```
    Announcements & Transfers
       (press PLAY or REC)

NUMBER  [425]   DELETE   BACK
```

- Press **NUMBER**

```
Type:   notification

OK      <<       >>      CANCEL
```

- Step through the message types with **<<** and **>>**
- Press **OK**.

```
Phone number: <unchanged>

NUMBER [425]   DELETE    BACK
```

This is the place to enter the telephone number that the Voice Sever will call while it plays the Direct Connect message or after it has spoken the Notification. The number can also be left as it is by default; unchanged. Then the Voice Server will connect to the number that was originally called.

Next you must define what you want to happen if the extension is busy or does not answer.

```
Busy/No Answer: < Off>

OK      [425]           CANCEL
```

You can fill in any message number: menu or transfer message (to another extension). If you want to rely on the procedure selected by the owner of the extension, (in this case the accountant) then you select “off” as the option for Busy/No Answer.

It is possible to have different procedures for busy and no answer. Just separate them with a slash, e.g.:

```
Busy/No Answer: 110/115

OK      [425]           CANCEL
```

8.2 Call Queue Transfer Messages

8.2.1 Introduction to Call Queues

A Call Queue is formed when a chosen number is busy. The telephone number can point to just one extension, but will usually point to a group of extensions of the PBX that will ring when a call to a that number is received. This is a standard function that is part of the configuration of the PBX. The maximum number of callers waiting in the queue is the same as the maximum number of channels in the Voice Server.

The queue is usually formed as a back up procedure of a normal transfer, Direct Connect or Notification. The reason is that normally the Voice Server will just connect the caller to the extension. Perhaps you want it to say something like: “ *You will be connected to our sales department”* (E.g. transfer message 405, that when the numbers are busy points to 406)
Alternatively you might leave the message empty, because then the caller will hear the phone ringing at once.

When the extensions are busy, the transfer message must point to a Queue message (e.g 406).

Callers placed in the queue will hear the transfer message e.g.:

"All employees of our sales department are busy, please wait a moment to be served." (E.g. transfer message 406)

Callers waiting will be transferred to the first free extension.

While waiting in the queue the caller can be informed about his progress in the queue. This will happen if the messages 050 to 071 are available.

A possible text for message 050 is:

"You are the first one in the queue to be served"

A possible text for message 051 is:

"There is one caller in the queue waiting before you"

A possible text for message 052 is:

"There are two callers in the queue waiting before you"

In between the Progress messages callers waiting in the queue will hear beeps at regular intervals. The beeps can be replaced by "music on hold". Music on hold can be installed by uploading a file named MUSIC.WAV to the Voice Server.

Please note that the call queue messages are not provided with the Voice Server. You must record them using the procedure to record voice prompts that we have already described.

There are two ways for the Anuncio to work with multiple extensions. It can keep track of the extensions that go off hook (System 1), or it can keep on trying the extensions (System 2). The difference between the two systems is that with System 2 the PBX may be annoyed by all the calls it gets because the Voice Server is trying to make the connection to the extensions that serve the queue. The two systems are functionally identical. System 1 has a drawback though: if the number of extensions that serve the queue changes, the configuration of the Voice Server must change too. If the number of extensions decreases while the Voice Server does not know it, System 1 will behave just like System 2. If the number of extensions increases without the Voice Server knowing, the extra extensions will not receive calls. Therefore we recommend that you use System 2.

When you are working on an extension that serves a queue, you will continuously receive calls. Since the PBX determines if you are member of the group (often called “hunt group”), you must tell the PBX when you want to stop. Otherwise the phone keeps ringing. Most PBX’s have a procedure or command to select or deselect the position in the hunt group from the extension. If you leave for a short time only, you may just take your telephone off-hook.

Unlike some other systems the Voice Server can have several call queues.

8.2.2 Configuration of the Call Queue

Configuration of a Call Queue is very similar to configuration of another Transfer message.

- Press the **Message** key and you will see:

```
   Announcements & Transfers
      (press PLAY or REC)

NEXT     [400]    CHANGE  STOP
```

- Record the message as you are used to do using the recorder keys, e.g.:

“All employees of our sales department are busy, please wait a moment to be served.”

- Press **CHANGE**

```
   Announcements & Transfers
      (press PLAY or REC)

NUMBER  [400]   DELETE   BACK
```

- Press **NUMBER**

```

Type:      Call Queue

OK        <<       >>    CANCEL
```

- Alter the message type of Call Queue with **<<** and **>>.**
- Press **OK**.

```
Phone number: <unchanged>
NUMBER    [400]    DELETE   BACK
```

The Phone Number can be filled in to be the DID number of the queue. However the likely way to come into the queue is from another transfer page that points to the number of the group that serves the queue. Therefore Phone Number can usually be left as “unchanged”.

```
Phones in Queue:    0
OK     <<          CLEAR    CANCEL
```

When the value is 0 the Anuncio will keep trying to put callers through. If a value between 1 and 30 is chosen, the Anuncio will not call the extensions until one of them is free. It will remember how many extensions are free and pass all incoming calls until the given number of calls has been put through and none of the extensions is free.

```
Busy/No Answer: 451
OK      [450]              CANCEL
```

The Busy/No Answer option of the Call Queue is different from that of other Transfer pages. As explained in the previous paragraph, the Call queues can be made to point to other Transfer pages.
The purpose is to program a hunt group. When the queue is trying to put a caller through, it can try a chain of transfer messages, usually without spoken content.

```
Busy/No Answer: <Off>
OK      [450]              CANCEL
```

Callers in the queue will remain in the queue. Even when the extensions are chosen by rotating through other Transfer pages.

8.2.3 More about Call Queues

First the audio file is played completely. After that the caller is placed at the end of a queue. Processing of the queue is dependant upon the specification of the "Phones in Queue" setting:

- **0** Auto mode: The voice server sends the telephone number to the PBX every 3 seconds. The PBX must return 'busy' if all telephones for this number are in use.
- **1-30** Represents the number of telephones available to answer calls on this queue. The Anuncio keeps track of the number of busy telephone lines and attempts to connect a caller when there are free lines or after another call ends.

Callers in a queue can be presented with audio messages announcing their position in the queue; music on hold; “On Hold Beeps” or one of the following combinations:

No system message 050 to 071 No audio file MUSIC.WAV	Caller hears “On hold beeps” at 5 sec. intervals
No system message 050 to 071 Audio file MUSIC.WAV present	Caller hears the Audio file until connected to the PBX. (Audio File should be long enough to last the entire waiting time)
Playback of one of the system messages 090 to 099	The System Messages 090 to 099 must be provided by the user. Otherwise identical to combination 2 above
System messages 050 to 071 present No play back of a system message specified	The appropriate position in the queue audio file is played when joining the queue. “On Hold Beeps” are then played as in combination 1 above until the caller’s position in the queue changes. The new position in the queue is played and “On Hold Beeps” heard once again. Finally the caller is connected to the PBX.
System messages 050 to 071 present System message 080 is specified	The appropriate position in the queue audio file is played when joining the queue followed by the audio file System Message 080.

	The next audio file between System Message 080 and 089 will be played when either the previous file is completed or the queue position changes. In the latter case the new queue position is heard first.

The Anuncio can process more then one queue. The telephone number distinguishes between them. The telephone number can be either the originally called number or a new telephone number specified for the queue. If the telephone number includes a "#" prefix the originally dialled number is retained if the caller escapes from the queue (see below).

There are several additional queue processing options:

- A "Queue Escape" Message Number may be specified. The caller can escape from the queue by pressing DTMF key 1 on their phone.
- A brief "Connected" Message Number can be specified. The audio file will be played when a the PBX signals a "Telephone in Queue" has been picked up (gone off-hook).
- A Queue Maximum can be defined. If the number of calls reaches this maximum, a jump is done to another message.

If no type is defined for a Transfer message, it will act as Direct Connect.

Transfer messages can optionally change the telephone number that will be sent to the PBX. The specified phone number can be a virtual number (only known by the Anuncio and the PBX) to enable interesting features to be created on the PBX. If no phone number is specified the originally called number of the incoming call is used.

For each Transfer message a Busy and/or No Answer message number can be entered.
If the dialled number is busy or doesn't answer before the "No Answer Timeout" time, the Anuncio can take back the call and jump to another message number. The No Answer Timeout is changed in the Anuncio's system menu.

Note1:
An “Auto” queue or busy-action is only possible if the PBX returns the busy-status, at the ISDN-level, if the number passed by the VS is occupied.

Note2:
Call queue requirements may be the same for multiple dial-in numbers (different called phone numbers). In this case duplicate queues would be required. A common queue message number is possible by specifying a queue number as the phone number. The queue number must be prefixed with a "#" ("#1" defines queue 1) and the number passed to the PBX is the originally called number.

9 System messages

System messages are fixed numbers from 000 to 099 that are used by the system for certain functions. The following numbers are in use:

000 - 009	*"zero" - "nine"* (spoken numbers, used to verify an input)
037	*"You have entered..."* (used after an input with verification option)
038	*"If ok press 1, to enter again press 2, to cancel press 3."*
039	*"One moment please..."* (used after an input when waiting for a response from the network)
040	Special Notification Message that can be included in recorded calls
042	Answering Machine Beep
043	On Hold Beeps used in the Call Queue
044	Phone Alerting Tone (ringing
045	Enable/Action Beeps used for various functions (high tones)
046	Disable Beeps (low tone)
050	*"You are the first person to be transferred"*
051	*"There is 1 person waiting before you"*
052 to 070	*"There are 2 (to 20) people waiting before you"*
071	*"There are more than 20 people waiting"*
077	*"You are now connected..."* (special transfer 'connect message' that is followed by ringing sound sm044)
080 to 089	Music-on-hold or advert files that are played sequentially, always starting with 080. These messages are played between queue position messages 050-070, replacing message number 043 (on hold beeps).
090 to 099	Music-on-hold sound files. If not present the system looks for the file **MUSIC.WAV**. The specified message is played instead off queue position messages 050-070. They should be long enough for the maximum time a caller will be held in the queue.

System Messages 50 to 71 are optional and used for the Call Queue. There are 22 message numbers available and there is only need to provide a maximum of the channels on that trunk or 8 for a BRI VS. The user records these messages using the suggested scripts, or similar, and copies them to the VS using FTP:

If the Call Queue messages are not present on the VS then the On Hold Beeps are used instead.

System Messages 80 to 99 are also optional and used only in a Call Queue. The user provides these messages and copies them to the VS using FTP. As with all cases of “broadcasting” music the user is responsible for the appropriate license:

10 Acknowledgements

10.1 Guarantee

Your Voice Server has a 12-month factory guarantee. The guarantee is effective for normal use only. We would like to emphasize that the guarantee is not valid under exceptional environmental conditions, such as extreme temperatures or humidity levels, nor in the unlikely event of a lightning strike. The guarantee is not valid if the machine has not been handled properly, for example when it has been dropped, or bumped into. In order to qualify for guarantee, you should contact your supplier, and show the receipt. If your supplier cannot help you, you should contact the manufacturer. The manufacturer reserves the right to determine the final date of the guarantee period on the basis of the date of production. Costs of transport to and from the supplier or the manufacturer are for the buyer's account. Guarantee is for parts only and does not cover any costs resulting from the breakdown of the Voice Server.

The Voice Server has various extra features that have not been described in this manual. Additional information about this is given in a technical information bulletin. Subjects discussed in this technical documentation are further configuration options, remote configuring, and how to update the firmware in the Voice Server. The further configuration options concern all aspects of operation. In our experience most people are interested in configurations related to user's access.

10.2 Liability

Correct functioning of the Voice Server cannot be guaranteed under all conditions and thus we do not accept any liability for loss of information or other damages due to the use of the Voice Server.

Vidicode is not a source of official interpretation of laws of any country or state and shall not be construed as a source for making decisions.

11 Appendix A Call Statistics

11.1 Example of the customer support department

A customer support department has 3 employees. Still your sales people are hearing from customers that support is often not answering the telephone.

In the PC software you make a query for all calls (incoming and outgoing) of the support people. You see that they have an average call time per day of 4 hours and 5 minutes per person. Therefore there probably is enough time for them to answer the support calls.

Then you investigate the type of calls they make and you see that many of them are incoming calls that take a lot of time, the average call time is 12 minutes.

You also make a chart of the distribution of the calls over the day. You see that the distribution of the calls over the day is not so equal. The majority of calls are received in the afternoon, the mornings are very quiet.

With the description above you have gained a good understanding of the situation. With the long calls mostly in the afternoon, it will be hard to get through to these support people, when you are calling yourself in the afternoon. Still the support people have quite some time left to serve more clients.

A possible solution is, to make a transfer message (possibly without announcement) on calls to the support group. Then if all extensions are busy, a call attendant menu will answer. The menu gives the caller the option to choose between a wait in the queue, or to leave a message to be called back the following workday in the morning. From the menu, the caller will either be transferred to the queue waiting for customer support, or a mailbox where he can leave his number. This will allow the customer to decide and will allow the department to be more effective because some calls can be placed in the morning.

It might be argued that call statistics are not that important in the given example, because the employees involved will be aware of the situation. Still the statistics form strong support for the solution because they provide accurate and unbiased information.

11.2 The example of many unanswered calls

When looking at the call statistics it can be seen that about 30% of all calls to the company remain unanswered. How to treat this? Is this the family of employees calling while the persons involved are not at their desk, or are these potential customers who now go to another supplier?

Call Statistics allow you to answer the question by making selections of local and remote numbers. So you refine your search to callers who have chosen the sales department from your main menu. You see that there is no problem here, everyone is served well. Then you make a query on the unanswered calls and you see that made have called sale people direct. The reason is that they know one person and that his DID number is on his business card. You may even sample a few remote numbers to verify if, after the call to a sales number was unanswered, they where taking the trouble to call the main number.

When you understand the situation a possible solution would be to make a transfer page on the DID number of every salesperson. The transfer page will monitor a call to the salesperson and, if unanswered or busy, connect to a menu for the sales group. This menu might offer to either leave a message to the person called or to speak to another salesperson. The command to speak to another salesperson will lead to a transfer page that calls all numbers in the sales group and forms a queue when they are all busy.

11.3 Call Statistics when the Voice Server is in service

When a Voice Server is in service, most calls will be answered. That however does not mean that all problems are solved. It is important to look at the call statistics a while after a procedure has been introduced. To give you an obvious example, it is very important to verify that callers do not hang up while they are waiting in the queue. If this happens it is even more important to know how long these callers have waited in the queue.

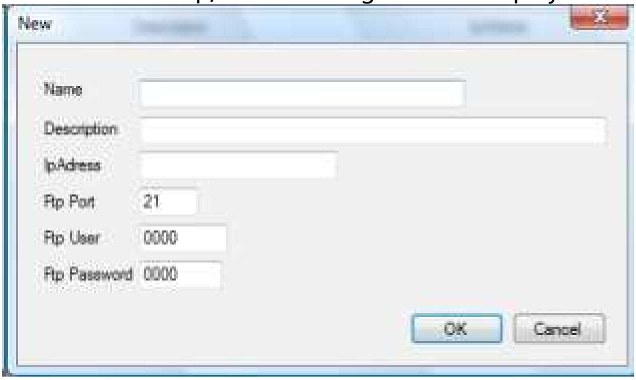
11.4 VC Reports software

Vidicode provides an application (VC Report) with which Call Statistics can be graphically displayed.
The software can be found on the installation CD. Locate the software as **Utils\vcReportInstall.exe.**
Run the program. The installation key is provided within the software.

11.4.1 Introduction to the report software

This software can be used to retrieve call statistics from Vidicode Voice Servers.

At the first startup, the following screen is displayed:



Fill in the name, IP-address, FTP username and password, and click OK. In the main screen, click the Refresh button to begin downloading the list of calls (menu Data, Refresh). This may take a long time, depending on the number of calls that are on the device.
In the main screen only a limited number of calls are displayed. (See menu Extra, Configuration, Settings to set how many). You can export the complete listing from the menu Data, Create Report.

From the menu Statistics, you can retrieve five types of statistics:

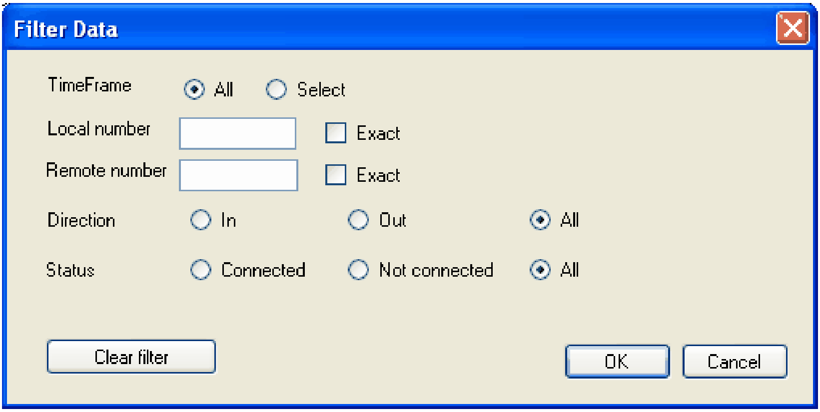
- Time Calls: The number of calls per time of the day
- Duration Calls: The number of calls per time duration
- Waiting Time: The number of calls per waiting time
- Number of Calls: The number of calls per day of the week
- Local number: The number of calls per local number

The statistics are available in three forms:

- Grid: Table-based information that can be exported to CSV format (for spreadsheet import)
- Chart: Graphical display
- Report: Text file

11.4.2 Filtering and searching

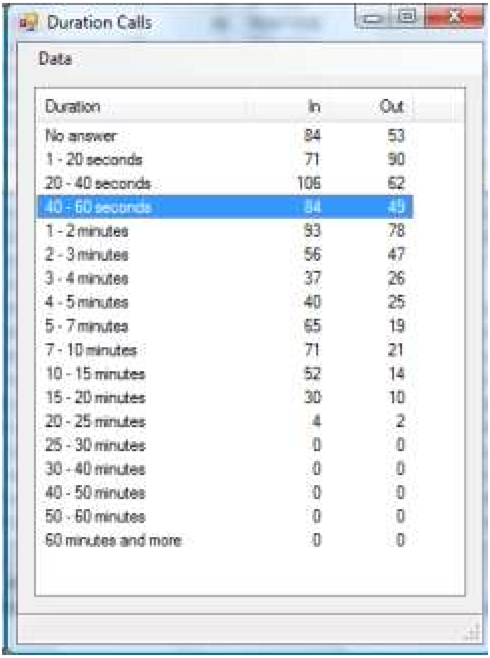
To filter the data to specific criteria, click menu Data, Filter Data. The data filter determines which calls are included in graphs, tables, and reports. The data filter is only a temporary filter and can be reset to include all calls.



When a local number is entered and Exact is not selected, the software will display all calls with a local number that contains the specified number. When Exact is selected, only calls with exactly the specified local number are displayed.
The clear filter button resets the filter. This has the effect that all calls will be included, after clicking on OK.

11.4.3 Statistics

There are three ways to view the statistics:
Grid

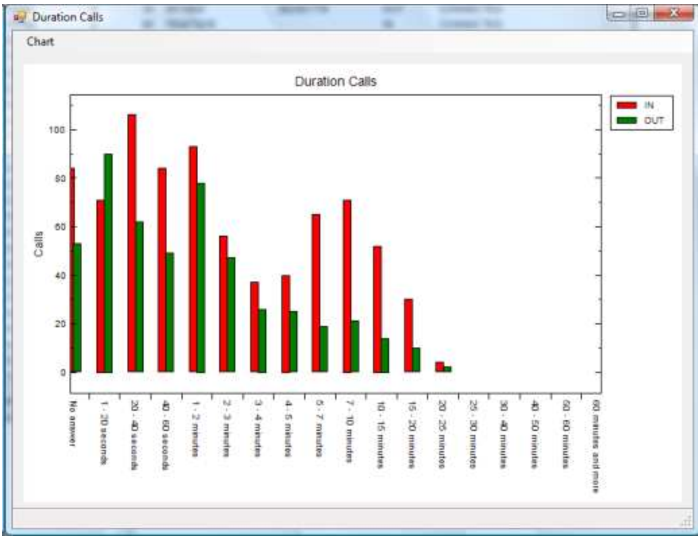


Duration Calls

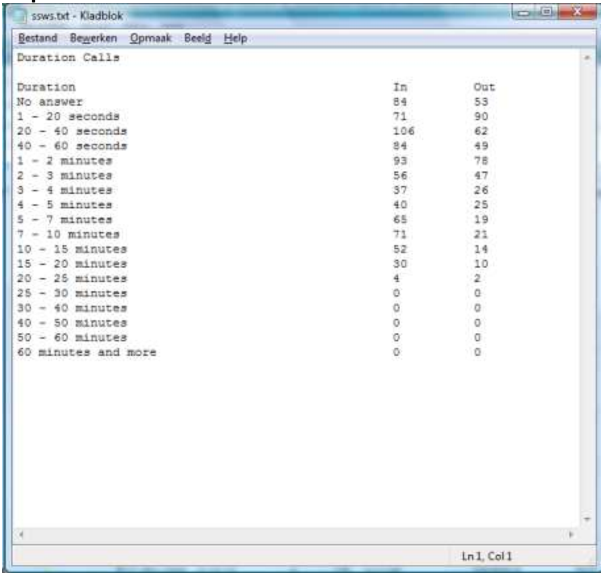
Data

Duration	In	Out
No answer	84	53
1 - 20 seconds	71	90
20 - 40 seconds	106	62
40 - 60 seconds	84	49
1 - 2 minutes	93	78
2 - 3 minutes	56	47
3 - 4 minutes	37	26
4 - 5 minutes	40	25
5 - 7 minutes	65	19
7 - 10 minutes	71	21
10 - 15 minutes	52	14
15 - 20 minutes	30	10
20 - 25 minutes	4	2
25 - 30 minutes	0	0
30 - 40 minutes	0	0
40 - 50 minutes	0	0
50 - 60 minutes	0	0
60 minutes and more	0	0

Chart



Report



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Duration Calls

Duration	In	Out
No answer	84	53
1 - 20 seconds	71	90
20 - 40 seconds	106	62
40 - 60 seconds	84	49
1 - 2 minutes	93	78
2 - 3 minutes	56	47
3 - 4 minutes	37	26
4 - 5 minutes	40	25
5 - 7 minutes	65	19
7 - 10 minutes	71	21
10 - 15 minutes	52	14
15 - 20 minutes	30	10
20 - 25 minutes	4	2
25 - 30 minutes	0	0
30 - 40 minutes	0	0
40 - 50 minutes	0	0
50 - 60 minutes	0	0
60 minutes and more	0	0

Ln 1, Col 1

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