TERAVoice Server 2004
TERASENS GmbH, the author(s), and any person or firm involved in the writing, editing, or production (collectively "Makers") of this manual ("the Work") do not guarantee or warrant the results to be obtained from the Work.

There is not guarantee of any kind, expressed or implied, regarding the Work, its contents and the product. The product and this manual are sold AS IS WITHOUT WARRANTY. You may have other legal rights, which vary from country to country.

In no event will Makers be liable to you for damages, including any loss of profits, lost savings, or other incidental or consequential damages arising from the Work, its contents and the product.

TERASENS GmbH
Ackermannstraße 3
80797 München
GERMANY
FON: +49 89 143370-0
FAX: +49 89 143370-22
E-Mail: info@terasens.de


Copyright © 2004 by TERASENS GmbH. All rights reserved. Printed in Germany. No part of this publication may be reproduced or distributed in any form or by any means, or stored in a database or retrieval system, without the prior written permission of the publisher.

Jegliche Arten der Vervielfältigung, Weitergabe und sonstige Arten der Verbreitung sind strengstens untersagt. Nachdruck, auch auszugsweise nur mit schriftlicher Genehmigung der TERASENS GmbH

Printed in Germany

All terms mentioned in this book that are known to be trademarks or service marks have been appropriately capitalized. TERASENS GmbH cannot attest to the accuracy of this information. Use of a term in this manual should not be regarded as affecting the validity of any trademark or service mark.

Microsoft and Windows are registered trademarks of Microsoft Corporation.
# Table of Contents

## 1 Getting Started

About TERAVoice ............................................................................................................. 12
- Unified Messaging ........................................................................................................... 12
- Benefits ........................................................................................................................... 12
- Hardware ....................................................................................................................... 12
- Flexibility ....................................................................................................................... 12
- Notification .................................................................................................................... 13

Using This Manual ........................................................................................................ 14
- Glossary ......................................................................................................................... 14

Features .................................................................................................................................. 15
- General ............................................................................................................................ 15
- Call Handler Assignment ............................................................................................... 15
- Call Handlers .................................................................................................................. 15
- Notification .................................................................................................................... 16
- TERAVoice Client Application ....................................................................................... 16
- Administration and monitoring ...................................................................................... 16
- Language support .......................................................................................................... 17

System Requirements .................................................................................................... 18
- Server .............................................................................................................................. 18
  - Supported Operating Systems ...................................................................................... 18
  - Hardware ..................................................................................................................... 18
  - Telephony hardware .................................................................................................... 18
  - H.323 Hardware (Voice over IP) ................................................................................... 18
- Client Tool ....................................................................................................................... 19
  - Supported Operating Systems ...................................................................................... 19

Hardware Compatibility List .............................................................................................. 20

Installation and Setup ....................................................................................................... 21
- How To Uninstall TERAVoice ......................................................................................... 21

Product Activation .............................................................................................................. 22
- Activation Methods ......................................................................................................... 22
- Online-Activation via Internet .......................................................................................... 22
- Offline-Activation via E-Mail ............................................................................................ 22

First Steps .......................................................................................................................... 24
- Telephony Device Configuration ..................................................................................... 24
- Notifications .................................................................................................................... 24
- Server Options ................................................................................................................ 24
- Setting up Call Handlers .................................................................................................. 24
- Starting TERAVoice .......................................................................................................... 25
- Making the first call ......................................................................................................... 25


2 Basic Concepts

Types of Telephony Applications ................................................................. 28
  Voicemail ........................................................................................................... 28
  Unified Messaging ............................................................................................ 28
  SMS and Pager Notifications ........................................................................... 29
  Scheduled Greetings ....................................................................................... 29
  Remote Control ............................................................................................... 29

Automatic Attendant ....................................................................................... 29

Voice Over IP Gateway .................................................................................. 30

Waiting Queues ............................................................................................... 31

Custom IVR Solutions .................................................................................... 31
  Programming Tasks ........................................................................................... 31
  Script Programming ......................................................................................... 32
  ActiveX Programming ..................................................................................... 32

Telephony Devices ......................................................................................... 33
  TAPI Devices ..................................................................................................... 33
  CAPI Devices ..................................................................................................... 34
  Device Features .................................................................................................. 34
  Feature support .................................................................................................. 34
  Device Groups .................................................................................................... 37

Call Handlers .................................................................................................... 38
  Voice Menu ......................................................................................................... 39
  Menu Events ....................................................................................................... 39
  Menu Actions ...................................................................................................... 40
  H.323 User .......................................................................................................... 40
    Incoming calls from the public network .......................................................... 40
    Outgoing calls from the IP network ................................................................. 40
    Authentication .................................................................................................. 41
  IVR Module .......................................................................................................... 41
  Music On Hold (MOH) ......................................................................................... 41
  Remote Control .................................................................................................... 42
    Authentication .................................................................................................. 42
  Time Schedule .................................................................................................... 42
    Actions ............................................................................................................... 43
    Unanswered Calls ............................................................................................ 43
  Waiting Queue ..................................................................................................... 43
    Cycle Modes ...................................................................................................... 43
    Queue Full Condition ....................................................................................... 44
    Announcements during wait ............................................................................. 44

Call Handler Assignment ................................................................................ 45
  Example ............................................................................................................... 45
  Matching Assignment entries ........................................................................... 45
  By Device ............................................................................................................ 46
    Device Groups .................................................................................................. 46
  By MSN/Extension .............................................................................................. 46
  By Redirector (Redirecting ID) .......................................................................... 47
  By Inband Signaling (DTMF Detection) ............................................................ 48
Table Of Contents

Network Configurations ........................................................................................................... 50
  Analog Networks .................................................................................................................. 50
  Analog Networks Without PBX ......................................................................................... 50
  Analog Networks With PBX ............................................................................................... 51
  Digital Networks .................................................................................................................. 52
  Digital Networks Without PBX ......................................................................................... 52
  Digital Networks With PBX ............................................................................................... 53

Miscellaneous Topics ............................................................................................................. 54
  Message Waiting Indication (MWI) ................................................................................... 54
  Text-To-Speech ................................................................................................................... 54
  TTS Engines ....................................................................................................................... 54
  Call Transfer Types ............................................................................................................ 55
    Native Call Transfer ....................................................................................................... 55
    Simulated Transfer ........................................................................................................ 56
    Simulated Hold ............................................................................................................... 57
  Sending SMS and Pager Notifications .............................................................................. 57
    SMS/Paging Services ..................................................................................................... 57
    Services and Routing ..................................................................................................... 57
  Language Support .............................................................................................................. 58
    Software User Interface Language ................................................................................ 58
    Voice Prompt Language ............................................................................................... 58
  Supported Audio Formats .................................................................................................. 59
  Security Considerations ..................................................................................................... 59

3 Using TERAVoice

Common Tasks ...................................................................................................................... 62
  How to set up Telephony Devices .................................................................................... 62
    Test Your Hardware ........................................................................................................ 62
    Configure Devices .......................................................................................................... 62
    CAPI Devices .................................................................................................................. 63
    Device Settings ............................................................................................................... 63
  How to set up Voice Mail .................................................................................................... 63
    Call Handler Assignment .............................................................................................. 63
    Voicemail Settings ........................................................................................................ 65
    Notifications .................................................................................................................. 66
    Setting up Mailboxes ..................................................................................................... 66
  How to set up Voice over IP Bridging (H.323) ................................................................. 67
    Steps to Activate ............................................................................................................ 67
    Setting up Clients .......................................................................................................... 68
  Using Remote Control from a Telephone ......................................................................... 70
    Configuring Remote Control ......................................................................................... 70
    Using Remote Control .................................................................................................. 71
  How to configure Inband Signaling .................................................................................. 71
    Activating Inband Signaling ......................................................................................... 72
    Setting up You PBX ....................................................................................................... 72
    Pattern Matching .......................................................................................................... 72
    Configuring Mailboxes ................................................................................................. 73
    Configuring Voice Prompts .......................................................................................... 73
4 Developing IVR Applications

Scripting API...................................................................................................................... 115
COM API ............................................................................................................................... 115
Support for .NET .................................................................................................................. 116
Comparison ......................................................................................................................... 116
### Table Of Contents

**Scripting API** .................................................................................................................. 117  
  General .......................................................................................................................... 117  
  Scripting Languages .................................................................................................. 117  
  Implementation .......................................................................................................... 117  
    Events ....................................................................................................................... 117  
    Properties and Functions ...................................................................................... 117  
  Steps to create an IVR Script ................................................................................... 117  
    Create the Call Handler ....................................................................................... 118  
    Configuring the Call Handler .............................................................................. 118  
    Creating the Script ............................................................................................. 118  
    Testing the Script ............................................................................................... 119  
    Implementing Error Handling ........................................................................... 119  
  Finding Errors ......................................................................................................... 120  
  Debugging Scripts ..................................................................................................... 120  
    Script Debugger .................................................................................................. 120  
    Enable Debugging .............................................................................................. 120  
    Debug Scripts ...................................................................................................... 120  

**COM API** .................................................................................................................... 122  
  General ....................................................................................................................... 122  
  Implementation .......................................................................................................... 122  
    Interface Details .................................................................................................. 122  
    Requirements ..................................................................................................... 122  
    Synchronization ................................................................................................. 123  
  Steps to create the COM Component ................................................................ 123  
    MS Visual Basic .................................................................................................. 123  
    MS Visual C++ 6.0 ............................................................................................... 124  
    MS Visual C++.NET 7 ........................................................................................ 125  
  Finding Errors ......................................................................................................... 126  
  Debugging Components .......................................................................................... 126  
    MS Visual Basic .................................................................................................. 127  
    MS Visual C++ .................................................................................................... 127  

**Script Editor** ............................................................................................................... 128  
  Menus ......................................................................................................................... 128  
    File Menu .............................................................................................................. 128  
    Edit Menu ............................................................................................................. 128  
    View Menu ........................................................................................................... 128  
    Tools Menu .......................................................................................................... 128  
  Toolbars ...................................................................................................................... 128  
    Standard Toolbar .............................................................................................. 128  
    Bookmarks Toolbar ............................................................................................ 129  

**Creating Outbound Calls** .......................................................................................... 130  

**IVR Samples** .............................................................................................................. 131  
  Scripting Samples .................................................................................................... 131  
    VBScript ............................................................................................................... 131  
    JScript ............................................................................................................... 131  
  Visual Basic Samples .............................................................................................. 132  
    Calculator .......................................................................................................... 132  
    Features .............................................................................................................. 132  
    Call Distribution ............................................................................................... 133
5 Troubleshooting and Support

Troubleshooting ............................................................................................................. 147
<table>
<thead>
<tr>
<th>Section</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>How to identify errors</td>
<td>147</td>
</tr>
<tr>
<td>Common Problems</td>
<td>147</td>
</tr>
<tr>
<td>Checklist</td>
<td>147</td>
</tr>
<tr>
<td>Event Log and Call Log</td>
<td>148</td>
</tr>
<tr>
<td>Common Problems</td>
<td>148</td>
</tr>
<tr>
<td>Activation fails</td>
<td>148</td>
</tr>
<tr>
<td>Check your telephony hardware</td>
<td>148</td>
</tr>
<tr>
<td>TERAVoice Does Not Start</td>
<td>150</td>
</tr>
<tr>
<td>TERAVoice Does Not Answer Calls</td>
<td>150</td>
</tr>
<tr>
<td>Product Support Options</td>
<td>152</td>
</tr>
<tr>
<td>Online Support</td>
<td>152</td>
</tr>
<tr>
<td>Product Updates</td>
<td>152</td>
</tr>
<tr>
<td>Knowledge Base</td>
<td>152</td>
</tr>
<tr>
<td>Newsgroups (Usenet)</td>
<td>152</td>
</tr>
<tr>
<td>Contacting TERASENS</td>
<td>153</td>
</tr>
<tr>
<td>Support by email</td>
<td>153</td>
</tr>
<tr>
<td>Support by Phone</td>
<td>153</td>
</tr>
<tr>
<td>Contact Information</td>
<td>153</td>
</tr>
</tbody>
</table>

**6  Glossary**

**7  INDEX**
1 GETTING STARTED

Welcome and thank you for using TERAVoice Server 2004, a new telephony server application that offers an unprecedented set of features and flexibility.

To help you get the most out of TERAVoice it is recommended that you read this documentation thoroughly and follow all hints and recommendations to ensure optimal results.

In order to get a quick overview of how this documentation is organized, please have a look at Nusing this Manual (p. 14).

This chapter will help you getting started quickly by summarizing the basic steps necessary to install and set up TERAVoice on your computer.

- About TERAVoice (p. 12) gives you a short introduction to the architecture of TERAVoice and which kind of applications and scenarios can be implemented with TERAVoice.

- For recommended and required hard- and software please refer to System Requirements (p. 18).

- A list of tested and supported telephony equipment can be found on the Hardware Compatibility List (p. 20).

- Installation And Setup (p. 21) tells you how to install TERAVoice Server on your system. Performing Product Activation is mandatory; otherwise TERAVoice will not be able to run.

- After you have installed the software you should follow the instructions given in the topic First Steps (p. 24) in order to get going.
About TERAVoice

TERAVoice Server 2004 is the new and universal telephony server platform by TERASENS. TERAVoice introduces new and unprecedented features incorporated into a single product. Scaleable from a single-line SOHO deployment up to a multi-server solution offering several E1 or T1 lines, TERAVoice is able to fit into virtually any environment.

UNIFIED MESSAGING

In a world of information and communication with increasing requirement in terms of productivity and flexibility, there is a growing demand for efficient messaging and communication solutions. The goal is to merge target all kinds of communication like e-mail, voice messages, fax, SMS, telex and others to a single target location, usually your e-mail inbox.

While solutions exist that try to offer all services in a single product, support for each single transport medium often lacks fully featured capabilities. TERAVoice, in contrast, is specialized in telephony services, providing flexible support for integration with other messaging systems and thus allowing customers to continue using the platform of their choice.

BENEFITS

TERAVoice can increase productivity by optimizing telephony communications. Never miss a call through instant notification on your mobile phone, pager or via email. Increase customer satisfaction through optimized hotlines with voice menus and managed waiting queues and create custom Interactive Voice Response (IVR) solutions in order to cut down on your cost for call agents.

With its MMC-based management console, TERAVoice can be administered and monitored easily, even from remote computers. For smaller configurations, no dedicated server is needed, simply plug-in an appropriate analog or ISDN telephony board into your file-server and get your telephony solution started at a very low cost, while maintaining scalability through being able to switch to a dedicated system for higher loads later.

HARDWARE

Through its integrated TAPI and CAPI interfaces TERAVoice is able to support a wide range of telephony hardware, starting from simple a voice modem up to Multi-PRI ISDN boards or professional TAPI-based telephony boards - even in mixed configurations.

FLEXIBILITY

TERAVoice combines the ease-of-use of simple voicemail solutions and the flexibility of programmable systems. Voice mailboxes, voice menus and several other Call Handlers can be created with just a few mouse-clicks. At the same time TERAVoice provides an easy to understand programming interface for creation of custom IVR application. Through its integrated scripting capabilities, building custom solutions is as easy as creating ASP (Active Server Pages) web pages.
NOTIFICATION

TERAVoice is compatible with all SMTP-based messaging systems and mail servers for delivering voice messages and call notifications. Additionally subscribers can be notified via SMS or pager messages and by the MWI (Message Waiting Indicator) of their PBX telephones. A windows client tool for notification and mailbox configuration is available as well.
Using This Manual

This documentation consists of five chapters describing several aspects of TERAVoice and of telephony applications in general. While it is recommended that you read through all parts of this document, this topic will help you find more quickly what you are interested in.

The current chapter, Getting Started, summarizes all steps necessary to get TERAVoice up and running without going into too much detail. This chapter is recommended reading for everyone installing and using TERAVoice for the first time.

In the chapter Basic Concepts (p. 27) you can read about general aspects of telephony applications, about the required technical infrastructure and how certain tasks are accomplished using TERAVoice.

The next chapter, Using TERAVoice (p. 61), deals with about system administration, configuration and setup. It describes all available configurable options and all tasks of system administration, including monitoring and error-tracking.

Tip: A lot of topics are covered both in Basic Concepts and in Using TERAVoice, where the first one gives a general explanation of specific features while the latter rather describes the configuration dialogs to configure these features. Links are provided in all those topics to easily switch between both versions.

Developing IVR Applications (p. 115) describes all options TERAVoice provides for the development of custom IVR applications. All provided programming samples are explained, including a reference to all API functions.

The last chapter, Troubleshooting and Support (p. 147), is all about identifying and resolving errors, with additional hints and links for getting further help. A list of common errors with possible solutions is included as well.

Glossary

Glossary terms are printed in bold italic. An explanation of these terms can be found at the end of this manual.
Features

GENERAL
TERAVoice is a high-performance multi-threaded windows system-service offering a unique and scalable telephony platform. The system can be easily expanded by purchasing additional licenses for lines and Call Handlers.

CALL HANDLER ASSIGNMENT
All incoming calls are handled by Call Handlers which offer specific call processing capabilities. The Call Handler that is designated to handle a specified call is determined by its Routing Parameters. TERAVoice offers four types of routing parameters:

- **By Device**
  This type of routing assigns the appropriate Call Handler depending on the telephony device or line on which the call occurs

- **By Inband Signaling (DTMF detection)**
  Most analog PBX systems can signal the station number from which a current call has been redirected to the telephony server by transmitting DTMF tones during the first few seconds of the call. This technique is also referred to as "In-band signaling"

- **By CalledID/MSN**
  This type of routing is typically used in digital networks. Telephony hardware can service different extensions (called MSN's - multi subscriber numbers in ISDN) on a single physical line. This MSN which is transmitted as the Called ID on TAPI devices can be used to assign a call to a distinct Call Handler

- **By Redirector**
  This type of routing is very often available in digital networks as well. If a call which is targeted to a certain extension is redirected to the telephony server (either by an explicit redirection the user has set on his telephone or by implicit redirection through PBX ringing groups) the PBX often transmits the original call target as the 'Redirecting Party Number'. This is an excellent option for easy mailbox assignment in digital networks.

CALL HANDLERS
The following types of Call Handlers are available in TERAVoice Server 2004:

- **Mailbox** (p. 39,96)
- **Voice Menu** (p. 39,98)
- **IVR Modules** (p. 41,101)
- **H.323 User (VoIP)** (p. 40,67,99)
- **Time Schedule** (p. 42,104)
- **Waiting Queue** (p. 43,105)
A detailed description of all Call Handlers can be found on page 38.

NOTIFICATION

Users can be notified about new calls or new messages with various options:

- **Email notification of new messages**
  An email message is sent to the configured email address each time a new message is recorded. An audio file of the message is attached to the email.

- **Email notification of new calls**
  For those calls where the caller does not leave a message a notification email can be sent to a different email address. This allows the subscriber to permanently keep up with all received calls. This function can be also used for creating call-logs in the messaging system. As an example a MS Exchange Public Folder’s email address can be used in order to create a call log in this folder.

- **SMS or pager notification**
  This method allows receiving notification messages on your mobile phone or pager either for all calls or only for those with a recorded message. The sender number for SMS messages can be set to the number of a remote control Call Handler for quick dialing into the remote control box.

- **MWI notification**
  Many PBX systems allow signaling new messages via the MWI-Indicator (Message Waiting Indicator) on their PBX telephones. This is done by dialing a certain special number for switching on and off.

TERAVOICE CLIENT APPLICATION

TERAVoice comes with a small Windows client application which loads in to the system tray. New messages are indicated via the tray icon and can be listened locally via the computer system’s speakers. This requires Windows network access to the TERAVoice server.

ADMINISTRATION AND MONITORING

TERAVoice can be administered and monitored locally and from remote systems. Administration is implemented as a Snap In for the Microsoft Management Console (MMC) and can be used like any other Windows administration tool.

An integrated monitor shows current activity with detailed call status information for all lines. Central logs for handled calls, informational and error messages are provided as well. TERAVoice can log information to Windows Event Log, TERAVoice internal log as well as a text file and even send error messages by e-mail to the administrator’s e-mail address.
LANGUAGE SUPPORT

TERAVoice currently offers support for both English and German language for windows user interface, documentation and all audio files. User interface language is depending on the current language setting of the currently installed operating system.

The audio interface language of system audio messages can be configured on a per-mailbox basis, in such way that user A can get English language remote control menu while user B can listen to the same menu message in German.

Further languages are planned and will become available for download as soon as available. Other languages for system voice messages can also be created by the customer.

Language support for voice menus or IVR modules can be implemented as desired.
System Requirements

SERVER

Supported Operating Systems
- Microsoft Windows 2003 Server
- Microsoft Windows XP Professional
- Microsoft Windows XP Home Edition
- Microsoft Windows 2000 Server
- Microsoft Windows 2000 Professional

Hardware
The required hardware depends very much on the maximum number of lines that need to be handled simultaneously and the other services that the server needs to perform. On modern server hardware that primarily serves as e.g. a file server TERAVoice usually runs without problem.

For systems with a larger number of lines a dedicated system should be used. Tests have shown that even older systems (e.g. Pentium 200 MHz) are able to serve a 2 line configuration, though.

Through its stable architecture for audio playback, a slow system performance usually results in a delay before audio playback starts instead of bad audio with glitches occurring.

Telephony hardware
TERAVoice offers support for an unprecedented range of telephony hardware. It supports both boards with TAPI (Microsoft Telephony API) and CAPI (Common ISDN API) interfaces. Most telephony boards come with their own TAPI driver while many European ISDN boards are shipped with a CAPI driver.

TERAVoice also supports the use of voice modems either through the Windows Unimodem/5 driver or a TAPI driver provided by the manufacturer so that low cost telephony solutions can be set up.

Please note: Be aware the not all telephony devices are suitable for every feature that TERAVoice offers. There are several features that require that a telephony device is capable of specific functions. Examples are call transfer, full-duplex support, 3party conference and many more. Please be sure that you choose a device that is really suited to your specific needs!

Some telephony devices that we have tested can be found in our Hardware Compatibility List (p. 20). If your device is not in this list it does not mean that it will not work but simply that it was not tested.

H.323 Hardware (Voice over IP)
TERAVoice allows bridging calls from and to IP network users via the H.323 protocol. While there is a lot of hardware available we did not test enough of these devices in order to create a compatibility list.
TERAVoice uses the Microsoft H.323 provider to connect to other H.323 terminals, so if you can connect a call to your H.323 device using the "Windows Phone Dialer" or "Microsoft Netmeeting", the device will be compatible with TERAVoice.

Users running one of those applications as their local phone are able to use the TERAVoice bridging and gateway capabilities anyway.

**CLIENT TOOL**

**Supported Operating Systems**

- Microsoft Windows 2003 Server
- Microsoft Windows XP Professional
- Microsoft Windows XP Home Edition
- Microsoft Windows 2000 Server
- Microsoft Windows 2000 Professional
- Microsoft Windows NT 4.0 Server
- Microsoft Windows NT 4.0 Workstation
- Microsoft Windows ME
- Microsoft Windows 98
Hardware Compatibility List

The hardware compatibility is constantly updated and not part of this documentation. The list can be found on the TERASENS web site under:

http://www.terasens.de/fwlink.asp?q=35010
Installation and Setup

Before you start installing TERAVoice you should check if all of the following is true:

- You have your original installation CD and your manual including license information and activation codes ready.
- You have not previously installed and activated this version on a different computer
- You already have installed all of the telephony hardware that you want to use with TERAVoice
- You have installed the latest drivers for your telephony hardware
- Your telephony hardware is connected to whatever kind of network you are using (see Network Configurations, p. 50)
- You have tested that your telephony device is able to connect to a remote party. (see Check your telephony hardware, p. 62)
- Now that you are ready to install, insert your TERAVoice CD and select "Install TERAVoice" in order to start setup.

Please enter your personal information and your serial number located on your User’s Manual or outside on the box of this package. (With electronic delivery you should have received this information via email.)

If you want to customize installation in order to change the installation path or to install the System Manager for remote administration please choose "Custom Setup".

During setup TERAVoice will try to detect if you are using any CAPI-based ISDN device. This is done by searching for the file CAPI2032.DLL. Depending on presence of this file support for those boards will be automatically selected or not. If you have a CAPI based board installed and you do not wish to use it with TERAVoice or if you want to use such kind of board which is not installed yet, you can manually change the installation state by selecting or de-selecting "Support for CAPI ". (You need to choose “Custom Setup” to get to this selection).

If you are satisfied with the selected options please click "Start Install" and wait until installation is finished.

Next step is to complete Product Activation (p. 22). Then proceed with First Steps (p. 24) to configure your system and prepare for the first start.

HOW TO UNINSTALL TERAVOICE

Please open Control Panel and click on Software. Scroll the list down until you can see the entry for TERAVoice Server and click on the Remove button.

Note: When uninstalling TERAVoice all Call Logs, recorded messages and user defined voice prompts will be retained on disk. If you do not want to keep this information you need to delete these files manually.
Product Activation

ACTIVATION METHODS

In order to protect our software against piracy TERAVoice uses a product activation mechanism similar to those found in other software packages.

Please note: TERAVoice will not run before it has been activated.

If you are unsure whether TERAVoice will fit your specific requirements, please download and test the free version of TERAVoice available for download from www.terasens.com. If you do not want to perform activation please return your package to the dealer where you purchased TERAVoice before performing activation.

While you can optionally supply your contact details for registration, activation itself is completely anonymous (if you are activating a license for the first time). There are two ways to complete activation:

ONLINE-ACTIVATION VIA INTERNET

Online-Activation is the easiest way for you to perform activation. All you need is an active internet connection and your license keys at hand.

Open TERAVoice Administration from your Windows Start menu, expand the node "Server Configuration", right-click on "Licensing" and select "Properties". The license dialog will show up.

Select "Add License" and enter your license key. If your license key is correct it will show up in the list with the status "not activated". Then click on "Activate License" and select "Online Activation".

You will then be asked if you want to activate and register or activate only. Registration is necessary if you want to obtain technical support from TERASENS. If you choose to register you need to enter your personal data on the next screen and press enter to confirm.

The software will now try to contact the TERAVoice license server via Internet. If the activation was successful the license will then show up in the list with the status "OK".

OFFLINE-ACTIVATION VIA E-MAIL

If you are unsure about what data is transmitted during online-activation or you do not have your TERAVoice system connected to the internet you can use email activation. To complete email activation, proceed as described in the previous paragraph but instead of online activation select "Email Activation", then choose "Create Request file".

You will then be asked if you want to activate and register or activate only. Registration is necessary if you want to obtain technical support from TERASENS. If you choose to register you need to enter your personal data on the next screen and press enter to confirm.

The software will then let you choose a location to save a file called tvLicReq.txt. Send this file via email to teravoice@registration.terasens.de. The TERASENS license server should usually respond within a few minutes. If activation was successful the response message will contain another attachment called tvLicResp.txt.
Now open TERAVoice Administration again and proceed to the license dialog. This time select "Process Response File" and specify path and location to the tvLicResp.txt file to complete activation.

The activation dialog is further explained in the topic Licensing and Activation (p. 90). If you have any problems with activation, please refer to the topic If Activation Fails (p. 148) in the chapter Troubleshooting And Support (p. 147).

As soon as you have completed activation you can continue with First Steps (p. 24).
First Steps

After you have successfully completed the Product Activation (p. 90) process it is now time to configure and start up TERAVoice. Start TERAVoice Administration from Start Menu ⇒ Programs ⇒ TERAVoice.

TELEPHONY DEVICE CONFIGURATION

At first you should configure your telephony devices. Expand the "Server Configuration" node, right click on "Telephony Hardware" and select "Properties". On the first tab you will find a list containing all devices that you can use with TERAVoice. Put a check next to all the devices you want to use with TERAVoice. More detailed information on device configuration can be found in the topic How To Set Up Telephony Devices (p. 62).

To configure the settings for CAPI based ISDN boards please select the device and press Configure. All those devices are named "ISDN Interface x ..." in the TERAVoice device list. For more information on using CAPI boards please refer to CAPI Devices (p. 63) and CAPI Configuration (p. 80).

The Next step is choosing which methods you want to use for Call Handler assignment i.e. how TERAVoice should determine what to do with a certain call. There are four different methods available in TERAVoice for this purpose, which are described in the topic Call Handler Assignment (p. 45). You can use any combination of those methods but at least one method must be used. In order to determine which methods are appropriate and available for your telephony configuration you should read the chapter Network Configurations (p. 50).

For a simple configuration or a first test you can select assignment method "By Device" and leave all others unchecked.

NOTIFICATIONS

If you are creating a voicemail solution you may want to set options how users are notified about new messages they have received. For a detailed description on how to set up notification options please refer to Notifications (p. 29,57).

SERVER OPTIONS

TERAVoice offers a lot of options for detailed customization of several features which are available if you right click on "Server Options" and select "Properties". More information about these settings can be found in the topic Server Options (p. 91).

The default settings can be used for a first test after installation.

SETTING UP CALL HANDLERS

Before you can finally start up TERAVoice you need to configure at least one Call Handler and set it as the default handler for all calls. For this purpose a mailbox handler is most suitable since it has a default audio file for playback.

If there is already set up a Call Handler of type 'Voice Mailbox' you can skip the following procedure:
Expand the "Call Handlers" node, right click on "Mailboxes", and then select "New Mailbox". Enter a name for your mailbox (e.g. "Default Mailbox") and press enter. The new mailbox now appears in the list in the right pane. Right click this new item and select "Set as default". The icon changes to a green arrow in order to indicate the default Call Candler. Please refer to Call Handler Assignment (p. 45) for further information about the default Call Handler.

**STARTING TERAVOICE**

Depending on whether TERAVoice is already running you either need to start or restart TERAVoice after changes to the configuration. Please refer to the topic When Is a Restart Necessary (p. 76) for further information.

In order to know whether TERAVoice is running you need to expand the "Monitor" node and select "Server Monitor". In the right pane you can see if TERAVoice is started and what the current status of operation is.

To start, stop or restart TERAVoice, please right click on "Server Monitor". If TERAVoice is started select "Restart", if TERAVoice is not running please select "Start" and wait until you can see the status change in the server monitor. If TERAVoice does not start please refer to the topic TERAVoice Does Not Start (p. 150).

**MAKING THE FIRST CALL**

After you have successfully configured and started TERAVoice you should now make your first call to the TERAVoice system. Call one of the devices you have selected and wait for the default mailbox to answer the call. If you cannot connect please go on with the topic TERAVoice Does Not Answer (p. 150).

If the mailbox answers the call with the default message you are done and have configured everything correctly. You can now continue to set up the Call Handlers necessary for your desired application. Please read the chapter Basic Concepts (p. 27) to get a better understanding of the options and the supported configurations. For detailed explanations on how to set up all available features please continue with the chapter Using TERAVoice (p. 61).
2 Basic Concepts

This chapter is an introduction to the basic concepts of telephony applications in general and the specific concepts used by TERAVoice to implement the supported types of applications.

Types of Telephony Applications (p. 28) describes what kind of applications and scenarios can be implemented with TERAVoice, what behavior and features are typical and what options TERAVoice provides for each type.

The second topic, Telephony Devices (p. 33), covers all aspects related to the hardware that can be used with TERAVoice, which drivers are needed and what functions a device needs to provide in order to be suitable for a desired purpose.

Calls are processed by Call Handlers (p. 38) in TERAVoice. There are several basic types of Call Handlers available. This topic describes each type and how it can be used.

In order to know which Call Handler TERAVoice should select for a certain call, there must be some assignment logic. Call Handler Assignment (p. 45) describes all available assignment criteria and how the assignment is handled by TERAVoice.

Depending on whether you are using a PBX or an analog or digital (ISDN) network, there are a lot of possible configurations. The topic Network Configurations (p. 50) describes those configurations and explains what kind of features can be used with each type of configuration.

Several other topics are covered in chapter Miscellaneous Topics (p. 54).
Types of Telephony Applications

Many types of telephony applications exist, some of them very common while others are rather specific, often to a degree that usual software can not supply the desired functionality out of the box. Therefore most telephony server applications provide a way of developing such applications by programming, scripting or other development procedures. Those applications are covered in topic Custom IVR Solutions (p. 31).

The more common application types are described in the following topics:

- **Voicemail** (p. 28)
  Provides an audio mailbox to one or more users on a network

- **Automatic Attendant** (p. 29)
  Provides secretary services by offering several options via voice menus, often depending on date and time

- **Voice Over IP Gateway (H.323)** (p. 30)
  Offers bridging of telephony calls to users on an IP network who are using an H.323 compatible (hardware- or software-) phone and vice versa

- **Waiting Queues** (p. 31)
  Call centers and workgroups often can not take every call immediately. Waiting queues can handle those calls, give the caller information and further instructions and finally connect him to a free agent as soon as one becomes available

- **Custom IVR Solutions** (p. 31)
  Developing custom telephony applications.

VOICEMAIL

Voicemail systems are an evolutionary step from the good old answering machine. If users are absent or busy their calls are diverted to an announcement of the user’s current availability, often with an option to leave a message. The storage of those messages is often referred to as "voice mailbox". Sometimes the maximum recording time is limited and an end greeting is played, sometimes mailboxes do not allow recording.

For a description of all steps needed to setup voicemail please refer to "How to setup voice mail" (p. 63).

Unified Messaging

Modern voicemail systems not only store messages but also offer functions to notify the mailbox owner about the new messages available. A common way of notification is an email sent to the mailbox owner, with the voice message attached to the email as an audio file. This allows the user to receive his or her mailbox messages as if they were normal email messages. This setup (often combined with the option to receive messages via other transport services like Fax, SMS, Telex, etc.) is often referred to as "Unified Messaging".

If you like to know how to set up voice mailboxes please jump to the topic Call Handlers - Mailbox.
**SMS and Pager Notifications**

Other types of notifications include notification via pager or via SMS to the user’s mobile phone in order for the user to be informed about new voice messages anytime and anywhere. For more information on how this notification is transmitted to the pager device or mobile phone, please refer to Sending SMS and Pager Notifications (p. 57). If you like to know how to configure notification options please read System Configuration - Notifications (p. 83).

**MWI Notification**

Some PBX systems also provide a special light on their PBX phones in order to indicate that there are new messages available in the mailbox. This is referred to as MWI (Message Waiting Indication) in PBX terminology.

Please read Setting MWI Indication (p. 54) for information about PBX systems implementing this functionality and System Configuration – Notification (p. 83) on how to set up MWI with TERAVoice.

**Scheduled Greetings**

Some mailboxes need different greeting messages depending on the time of day or on the current date or weekday. A user might, for example, want to specify a special greeting to be played during his vacation or a corporate mailbox should play a different message during business hours (e.g. weekdays from 0900h until 1700h) than outside of business hours. Please go to Call Handlers – Mailbox for information on how to configure scheduled greetings.

**Remote Control**

Usually the most important way to listen to messages that have arrived is remote control. Remote control access to mailboxes can be achieved by several methods. One way is to have the user call his own mailbox and press a certain key on his keypad in order to switch to remote control mode; then a PIN number has to be entered in order to authenticate the user and allow remote control.

Another method is to provide a central number which all users call in order to gain remote control access. With this configuration the user has to enter his mailbox ID in addition to the PIN code, because the system does not know which mailbox the user wants to get remote control access to.

If the telephony infrastructure allows transmission of caller ID information the caller ID can be used to implement authentication without entering mailbox ID and PIN numbers. For information on how to configure remote control settings for TERAVoice please read Remote Control (p. 42).

The voice menus for remote control mode support different languages. For more information about language support in TERAVoice please refer to Language Support (p. 17,58).

**AUTOMATIC ATTENDANT**

Many offices employ assistants for telephony services with highly repetitive tasks like transferring calls to either sales or support or telling customers about business hours and availability of certain departments. Many of those tasks can be solved by so called automatic attendants which are implemented by voice menus. An example for such an attendant could be:
"Hello and welcome to TERASENS Corporation. Please press 1 for our domestic sales department, press 2 for international sales, press 4 to get connected to our support line, press 5 if you know the extension of the person you want to get connected to. If you need further help or do not know what option to choose, please press 0 and we will connect you to our telephone exchange."

If the call occurs outside usual business hours the following message could be played:

"Hello and welcome to TERASENS Corporation. Our business hours are Monday to Friday from 9:00am to 7:00pm. If you want to leave a message please press 1, if you want to get further information on how to contact us by email or fax please press 9. Thank you for calling!"

This functionality can be implemented in TERAVoice by a mix of several features:

- **Play messages and perform actions upon digit input:** Voice Menus. Read Call Handlers -- Voice Menu (p. 39) for basic explanation and Setting up Call Handlers -- Voice Menu (p. 98) for instructions on how to configure them.

- **Play messages depending on weekday, date and time:** See Call Handlers -- Time Schedule (p. 42) for basic explanation and Setting Up Call Handlers -- Time Schedule (p. 104) for instructions how on to configure these.

- **Transfer a call depending on the extension received via digit input:** Refer to Custom IVR applications (p. 31) for information how to create a script to play a voice prompt to enter the extension number, check extension for validity and finally transfer the call.

**VOICE OVER IP GATEWAY**

Telephony connections that are connected via VoIP (Voice over IP) offer a cost-effective solution compared to legacy telephony devices i.e. telephones. The modern Windows operating systems even come with two software applications implementing IP telephony via the H.323 protocol: The "Windows Phone Dialer" and "Microsoft Netmeeting". There are other applications that implement a phone in software as well as hardware telephones that can be used on an IP network. While communication with other devices on the same network works "out of the box", connecting to other parties on a public telephony network requires a gateway between the IP network and the telephony network.

TERAVoice provides the required gateway functionality through its ability to bridge outgoing calls from the IP network to the public network and connect calls from the public network to users on the local IP network.

Every user on the local IP network has to be configured as a Call Handler in TERAVoice and thus can be assigned to certain inbound calls like any other Call Handler (see Call Handler Assignment, p. 45).

For outgoing calls to work the server running TERAVoice must be configured as the H.323 gatekeeper on each user’s computer. All H.323 calls will then be made to the TERAVoice server regardless of the destination address. If the destination address is of in the form of a phone number, TERAVoice will try to establish a connection to the desired remote party and bridge the call as soon as the external call gets connected. For other types of destination numbers like computer name or IP address, TERAVoice diverts the call directly to the target or to a different gatekeeper that handles calls over the internet through a firewall (like the H.323 gatekeeper of Microsoft ISA Server).

For outgoing calls TERAVoice checks if a Call Handler for the desired internal user exists, so only identified users can connect outgoing calls. If the TAPI device supports that functionality, TERAVoice can set the outgoing caller ID to a value corresponding to the internal user.
For detailed steps to configure the H.323 gateway function please refer to How to set up Voice over IP Bridging (p. 67).

**WAITING QUEUES**

Usually no modern call center can guarantee the availability of a human agent to handle an incoming call at all times. Callers therefore need to be informed that it could take some time until their call will get connected. Calls usually need to be distributed evenly among all agents.

During their waiting period callers can be informed about news, hints and tips or frequently asked questions in order to encourage self help and to minimize the use of call center resources.

To improve acceptance and customer satisfaction it is often desired to inform the caller about how long he will need to wait. This could be done either by informing the caller about the average waiting time during the last 15 minutes or by announcing his current queue position each time his position changes.

Queues can be limited to a certain amount of slots in order to prevent waiting times which are unacceptable for the caller. A special action can be configured to perform on calls when all queue slots are already occupied. An example would be an action that transfers the call to another department or disconnects the call after playing a message that the caller should try again at a later time.

As soon as the caller has reached position one in the queue the telephony server application tries to find a free agent and transfers the call. Several options are available for call transfer. For a detailed description of transfer modes please refer to Call Transfer Types (p. 35,55).

**Call Recording**

Some call centers need to record all agent conversations. Recording those calls is available in combination with transfer mode "3 party conference", if the PBX or the telephony network supports this feature.

For more information on configuring waiting queues please read topic Setting up Call Handlers -- Waiting Queues (p. 105).

**CUSTOM IVR SOLUTIONS**

While certain types of applications are very common so that standard software can offer the required functions out of the box, a lot of scenarios are so specific that the functionality must be implemented by some type of custom programming.

**Programming Tasks**

There are several tasks that a programming environment needs to support when interfacing with the telephony server:

- Playing back audio
- Recording audio
- Processing received digits
- Forwarding processing to a different Call Handler
- Transferring the call
- etc.
The programming environment needs to be able to perform other functions depending on the type of application, for example:

- Accessing, reading and updating databases
- Interfacing with other applications like web servers or mail servers
- Accessing other systems, e.g. for retrieval of ordering information etc.
- And many more

TERAVoice offers several methods to implement this type of functionality. The easiest but least flexible option without ability for programming is to use Voice Menus.

**Script Programming**

The first method for programming IVR solutions with TERAVoice is using scripts. While TERAVoice officially only supports VBScript and JScript, any scripting language that can be installed for the Windows Scripting Host can be used with TERAVoice (e.g. Perl, Python etc.)

**ActiveX Programming**

A second method for programming with TERAVoice is creating ActiveX (in-process) components. The components need to implement a certain COM interface.

Any type of programming language can be used for this purpose as long as it is able to create polymorphic components (i.e. components with more than one interface). Examples of suitable programming environments are Microsoft Visual Basic or Microsoft Visual C++.

**Event Programming**

The programming model is the same for scripting and ActiveX objects. The script or components receives a number of events to which it needs to respond:

- Initialize is called when Call Handler is loaded
- UnInitialize is called when Call Handler is unloaded
- DigitReceived is called for every digit the caller has pressed
- FileSaved is called whenever a recording was completed
- PlayDone is called when playback of an audio file was completed
- TimePulse is the most important event: It is called at least every second and additionally after any status change of playback or of call state etc.

A recommended concept for programming applications with this interface architecture is to program simple state machines. Global variables keep track of the current state of the call which can be changed during events as needed. This method combined with the fact that the TimePulse event is fired repetitively even when no status change occurs, provides the greatest possible flexibility for creating all types of applications.

More information on configuring Script or ActiveX handlers can be found in the topic **IVR-Module** (p. 41). For detailed help on creating IVR solutions with programming samples please refer to **Developing IVR Applications** (p. 115).
Telephony Devices

During the last decade two different types of telephony API (Application Programming Interface) have developed on the Microsoft Windows platform. TAPI (Telephony API) – a standard developed by Microsoft – is part of the Windows operating system and can be considered as standard for most telephony devices around the world. CAPI (Common ISDN API) is a standard of the CAPI organization (www.capi.org) and mainly used for ISDN boards on the European market. Previously customers had to choose between the two standards and select a telephony software solution for the selected API. With TERAVoice being able to handle both APIs on a single platform, customers are no longer forced to make this kind of decision. Devices of each kind can be used simultaneously with no restrictions. The specifics of each API are described in the following topics:

- TAPI Devices
- CAPI Devices

Since there are a lot of devices suited for specific environments there are sometimes great differences in the functions supported by these devices. Several functions in TERAVoice require that the telephony devices offer certain features. For more information about these device features read Device Features (p. 34).

Devices can be configured into Device Groups (p. 37) for easier management and selection.

TAPI DEVICES

TAPI is the Microsoft windows default API for telephony applications. TAPI drivers are available for most telephony devices on the market. Since TAPI defines a very flexible interface, several types of devices exist which are not all well suited for telephony server applications like TERAVoice. For example some devices only support assisted telephony which means that the computer can dial a number on an external device like a telephone. Other TAPI drivers can interface with PBX systems for call control and automatic call distribution.

TERAVoice requires devices that support the interactive voice media mode, which means that the device needs to be able to transmit and receive audio data. These devices are commonly referred to as voice boards or telephony boards. Not all boards are equally well suited to be used with TERAVoice. Some manufacturers do not place a high priority on TAPI driver development, which sometimes results in poor TAPI driver implementations. For a list of drivers that were successfully tested with TERAVoice please refer to our Hardware Compatibility List (p. 20).

Voice modems (Unimodem/5)

A special case of TAPI driver is the Windows Unimodem/5 driver, which supports most modems under Windows. If a modem is capable of voice mode it can be used with TERAVoice. Several restrictions apply here, though. Most modems offer really poor support of voice mode. A list of some modems that were tested with TERAVoice can be found on our Hardware Compatibility List (p. 20).

Additionally Unimodem/5 only supports half duplex operation, which for example prevents H.323 gateway functions. Several other features are not available with voice modems, too. For a more detailed list of device features and information which features are required for certain TERAVoice functions please refer to the topic Device Features (p. 34).

In general the use of voice modems can only be recommended for simple one-line applications that do not require high audio quality or specific features like call transfer or voice over IP bridging.
Procedure where A calls B. B creates a consultation call to C while holding A. B then connects/transfer A to C.

CAPI DEVICES

CAPI (Common ISDN API) is a standard of the CAPI organization (www.capi.org) and is mainly used for ISDN boards on the European market. The use of CAPI boards is enabled via the ComISDN TAPI for CAPI driver which is included with TERAVoice.

All features of TERAVoice can be used with CAPI based boards if the board’s CAPI driver and the telephony network or the PBX supports the required functions.

For a more detailed list of device features and information which features are required for certain TERAVoice functions please refer to the next topic Device Features (p. 34).

Extended services that are most commonly provided by ISDN networks and which can be used with TERAVoice are Multi Subscriber Numbers (MSN), Call Transfer, Call Hold, 3 Party Conference, redirector information as well as several others.

DEVICE FEATURES

Feature support

The following table provides a common overview of feature support with different driver architectures. For detailed information about supported devices and features please refer to our Hardware Compatibility List (p. 20).

<table>
<thead>
<tr>
<th>Feature</th>
<th>Common TAPI</th>
<th>Unimodem/5</th>
<th>CAPI</th>
</tr>
</thead>
<tbody>
<tr>
<td>DTMF Detection</td>
<td>Supported by most devices that support audio</td>
<td>+</td>
<td>+</td>
</tr>
<tr>
<td>Full Duplex</td>
<td>Supported by some voice boards</td>
<td>-</td>
<td>+</td>
</tr>
<tr>
<td>Caller ID indication</td>
<td>Depending on device (depends on modem)</td>
<td>+</td>
<td>+</td>
</tr>
<tr>
<td>Call Transfer</td>
<td>Depending on device</td>
<td>-</td>
<td>+</td>
</tr>
<tr>
<td>Blind Transfer</td>
<td>Depending on device</td>
<td>-</td>
<td>+</td>
</tr>
<tr>
<td>3 Party conference</td>
<td>Depending on device</td>
<td>-</td>
<td>+</td>
</tr>
<tr>
<td>Call Hold</td>
<td>Depending on device</td>
<td>-</td>
<td>+</td>
</tr>
<tr>
<td>Called ID (multiple MSNs or extensions per device)</td>
<td>Depending on device¹</td>
<td>-</td>
<td>+</td>
</tr>
<tr>
<td>Redirecting ID indication</td>
<td>Depending on device¹</td>
<td>-</td>
<td>+</td>
</tr>
<tr>
<td>Hangup detection</td>
<td>Depending on device</td>
<td>Sometimes not reliable</td>
<td>+</td>
</tr>
<tr>
<td>Call progress detection (outgoing calls)</td>
<td>Depending on device/nginx</td>
<td>-</td>
<td>+</td>
</tr>
</tbody>
</table>

¹usually only available on digital networks or with PBX specific boards
**DTMF Detection**
Ability to detect a caller’s key presses on the telephone keypad. This feature is required in all situations where an input of the caller must be processed, such as voice menus, remote control and other IVR applications.

**Full Duplex Mode**
Full duplex mode allows simultaneous transfer of audio data in both incoming and outgoing directions. Full duplex mode is not required for voicemail or other IVR applications since the software decides when to play an announcement or record a message. DTMF digit detection works in half duplex mode as well, so that digits can be processed while playing back audio. Full duplex mode is required when calls are being bridged to or from H.323 users and when using software simulated call transfer ("Bridged Mode").

Almost all professional telephony boards and all CAPI boards support full duplex audio. Unimodem/5 driven voice modems never support full duplex mode.

**Caller ID indication**
In some analog networks and in most digital telephony networks the number of the calling party (Caller ID) can be transmitted to the called party during ringing. The Caller ID is used for display in call logs, for user authentication in remote control modes and can also be used for further processing in custom IVR applications like looking up and forwarding the call to the employee or department assigned to the customer.

Most analog and digital devices support transmission of Caller ID information if supported by the network. Even some (but not all) Unimodem/5 driven voice modems support Caller ID information.

**Call Transfer**
Call Transfer is mainly used in combination with PBX systems when a call needs to be handed over to another employee. Example: A calls B. Without disconnecting, B calls C and then executes the transfer. As a result, A is connected to C, B is not connected to anyone anymore.

TAPI defines this function as a two step process: First a consultation call is being made. If that call gets connected, the transfer can be completed.

This function is available with analog and digital PBX systems as well as with some public digital networks.

In ISDN networks there is a standardized procedure for Call Transfer. All ISDN devices (TAPI or CAPI based) that support Call Transfer should be able to transfer calls if the network or PBX supports it.

With analog PBX systems the procedures for call transfer vary. Usually a hook/flash is sent on the line followed by the extension number to connect the consultation call to. After connecting the consultation call the transfer can be completed by hanging up. The actual sequence may differ, though. TERAVoice is not capable of executing such digit sequences; instead it calls the TAPI transfer functions in order to execute a Call Transfer. Professional analog telephony boards allow defining those sequences which are in executed if a TAPI application calls the mentioned functions.

Unimodem/5 voice modems are not able to 'learn' those transfer sequences and do not support the required transfer functions, thus call transfer is not available with those devices.

**Blind Transfer**
Blind transfer is a special TAPI function that performs Call Transfer as a single-step process. Applications have less control over the transfer progress and cannot abort the transfer process if the consultation call does not get connected.

Blind Transfer mode is only included as a selectable mode in TERAVoice because there are some TAPI drivers that only support Blind Transfer.

All other information mentioned for Call Transfer applies to this feature, too.
3-Party conference
3-Party conference can be considered as a call transfer where the second party does not get disconnected: A calls B. B calls C and initiates a conference. All three parties can hear each other. With TERAVoice this feature is never used for a 3-party conference with three actual callers. Nevertheless there are two special cases in which this feature is useful:

- You want to transfer calls but your public network supports only conference but not Call Transfer and you do not have a PBX. (All PBXs that support conference, support transfer as well, so this should be only interesting if you are directly connected to a public network)
- Of course you could also use simulated transfer (bridged mode) but you can save one line compared to bridged mode because a conference only needs one line
- You are servicing your call center with a Waiting Queue Call Handler and you want to record calls. In this case you can use conference instead of transfer in order allow TERAVoice to record the conversation

3-Party conference is usually only supported on ISDN networks.
Not available with Unimodem/5 modems.

Call Hold
Call Hold interrupts the connection between the caller and the called user or application without actually disconnecting the call. The caller hears a tone signal or some music being played which is called Music on Hold (p. 41,101) (TERAVoice can also be used to feed the hold music into a PBX via its Music on Hold Call Handler). The call can be fetched back from hold state any time. When doing Call Transfer the first call is always put on hold while the consultation call is being made. Call Hold can be used with Waiting Queues in TERAVoice in order to have the caller hear some music or interesting information while waiting to get connected. TERAVoice can also simulate hold by directly playing hold music.

This feature is available with some TAPI and all CAPI devices. Not available with Unimodem/5 modems.

Called ID
For telephony server applications it is very useful if the device can operate on different numbers, so that several different services can be handled by a single device. In ISDN networks a device can be assigned several MSN's (Multi Subscriber Numbers). Some PBX specific boards may also provide a similar function. In order for the application to know which extension (or complete numbers if not connected to a PBX) was called, this information is transmitted as the Called ID. More information can be found in Call Handler - Assignment – By MSN/Extension (p. 46).

Called ID is available with some TAPI and all CAPI devices; it is not available with Unimodem/5 modems.

Redirecting ID indication
When a user A calls to B and B has set a call diversion to C the call from A will be redirected from A to C. C will receive the phone number of A as the Caller ID. If the network supports that feature, C will receive the number of B as the redirector which is called the Redirector ID. This redirector information can be used for example in voicemail applications to determine the mailbox that a call is destined for. For more information refer to Call Handler Assignment – By Redirector (p. 47).

Usually this information is only available in ISDN networks. While only some TAPI based ISDN boards might support this feature most CAPI boards report this information.

Not available with Unimodem/5 modems.

Hang-up detection
While detection of hang up is very easy in digital networks where an explicit message about disconnection is sent, in analog networks this is sometimes more difficult than it seems.
Depending on the network sometimes special tones are sent in order upon disconnection of the other party while sometimes special DSP processing or detection of impedance changes is necessary to detect disconnection reliably. Professional analog voice boards provide those functions while Unimodem/5 modems sometimes have difficulties in correctly detecting hang-up. This could for example lead to minutes of recorded audio for voicemail applications while the other party has already disconnected after a few seconds.

Currently TERAVoice does not support any methods of silence detection to prevent this problem because TERAVoice is focused on professional use and does not recommend the use of voice modems. Some combinations of modem and network/PBX may work; some may not work reliably in this regard.

**DEVICE GROUPS**

The TAPI drivers that come with your telephony hardware can use different driver models. Some drivers let you choose between the available modes while some do not. The main difference is the relation of TAPI devices and lines.

**Example:** The driver that comes with a T1 telephony board could either install a single device (in the TAPI list of devices) that operates 24 lines, or it could add 24 devices to the TAPI device list each of which operates a single line.

In order to simplify management and Call Handler Assignment (p. 45,63,78) with a large number of devices, TERAVoice provides a feature for creating and assigning devices to device groups. This method is also useful if you are using several telephony boards (whether of the same type or of different types). Details about creating and managing device groups can be found in the topic Device Groups (p. 37).
Call Handlers

In TERAVoice the call processing functionality is organized into Call Handlers. Call Handlers determine the type of processing that is going to be applied to a call. Any incoming call has to be initially assigned a Call Handler. The mechanisms that are used to assign a Call Handler to a call are described in Call Handler Assignment (p. 45).

During call handling some types of Call Handlers allow forwarding processing to another Call Handler. For example a complex menu structure can be built with several voice menus that forward processing to other voice menus or IVR Modules depending on user input.

MAILBOX

The Mailbox Call Handler defines a mailbox for a user or an organizational mailbox. There are several ways in which calls targeted to a certain user can reach the telephony server.

Redirection to Mailbox

Usually a user redirects calls from his phone to the mailbox if he is out of office or does not want to be disturbed. With most PBXs the call can be redirected unconditionally (immediate) or after a delay. The delayed redirection should always be enabled while the unconditional redirection can be activated and deactivated as desired. In the later case a mailbox in TERAVoice would be configured to answer calls immediately.

Some PBXs allow configuration of ringing groups. If a user does not pick up the phone after a certain time, the call is redirected to the members of this group. This gives other users which belong to this ringing group the opportunity to take the call. TERAVoice could in this case be configured to be part of this ringing group and take the call after a delay. In order for TERAVoice to know which user the call was originally targeted to, it is required that the PBX transmits that number as the Redirecting ID. For more information please refer to Call Handler Assignment (p. 45).

TERAVoice can also be set up in parallel to the user’s phone and take the call after a preconfigured delay. This configuration is primarily used for configurations without a PBX.

Announcements

After the preconfigured duration the call is taken and an announcement is played back. More than one announcement can be configured depending on the day of the week, the date and time in order to allow special messages during a user’s holiday or for organizational mailboxes to play different messages during and outside of office hours. A mailbox can be set to allow recording a message of a configurable maximum length; after the configured duration recording will be stopped and a configurable goodbye message can be played if desired.

Notifications

In order to always stay informed about new messages that have arrived several types of notification options are provided.

The most common method is to send an email to the user’s email account with all details of the call and the message attached to the e-mail as an audio file, if a message was recorded.
Another way of notifying a user is to send a message to his or her mobile phone or pager. Some PBX systems offer an indicator light on their phones to signal that new messages are available.

For more information about notification options please refer to Message Waiting Indication (p. 54) and Sending SMS and Pager Notifications (p. 57).

**Listen to new messages**

TERAVoice provides three methods for listening to new voice messages for voicemail users:

- Receiving voice message as an audio file via e-mail
- Listening to voice messages using the TERAVoice Client Application
- Using Remote Control

When using Remote Control the user calls his voice mailbox or an extension with a Remote Control Call Handler (p. 103) and dials a certain code on his telephone’s touch-tone keypad in order to enter Remote Control mode.

**Remote Control**

In order to allow remote configuration of mailbox settings - for example changing the default announcement or activating and deactivating a mailbox - as well as for listening to new messages in the mailbox a remote control mode is required. Remote control can be invoked directly from a mailbox by pressing ‘#’ and entering a PIN number or by calling a number to which a Call Handler of the type Remote Control (p. 42) is assigned. For more information about remote control mode please proceed to Call Handlers - Remote Control (p. 103). The menu language for remote control can be customized as described in Language Support (p. 58).

Information on how to set up and configure mailboxes can be found in Setting up Call Handlers – Mailbox (p. 66).

**VOICE MENU**

Voice Menus offer any easy way to implement simple IVR functionality. Voice Menus simply consist of a message that is being played and several actions that can be carried out.

**Menu Events**

Two types of events are available for voice menus:

- Digit Input events
- Timeout events

Digit Input events occur when the user presses a key on his touch tone keypad. For each key a different action can be configured. If not all keys are being used, an event can be defined that occurs for ‘all other digits’.

Timeout events occur after a specified time, so that a specific action can be carried out if the user does not select anything.
Menu Actions

The following actions can be configured to be carried out on the selected events:

- Repeat message
  - Repeats the menu announcement
- Transfer call
  - Transfers the call to another phone number
- Forward to Call Handler
  - Forwards the call to a different Call Handler
- Hangup Call
  - Disconnects the call

For information about Call Transfer please read Call Transfer Types (p. 55). For instructions how to configure Voice Menus, please refer to Setting up Call Handlers – Voice Menu (p. 98).

H.323 USER

IP telephony is becoming more and more popular. Modern operating systems usually ship with an integrated H.323 telephony application which lets a user connect to other users on the local LAN or the internet. Hardware devices which can be plugged directly into the local LAN are also available.

IP telephony offers a much cheaper telephony solution in many cases since no dual wiring is needed for telephony and LAN and the computer with a connected headset can be used instead of a telephone. Obviously this type of configuration does not make much sense if subscribers of the public telephony network cannot be reached.

TERAVoice provides the functionality for reaching the public telephony network by acting as a gateway between H.323 based VoIP networks and a public telephony network (or a PBX).

For each user on the IP network who wants to use the H.323 gateway functionality a Call Handler of the type H.323 User needs to be configured,

Incoming calls from the public network

For incoming calls the H.323 Call Handler can be assigned to certain incoming calls like any other Call Handler. Other Call Handlers can forward processing to a H.323 user just the same as to any other.

Upon activation TERAVoice tries to connect the call to the user on the IP network via H.323. As soon as the call is being taken the initial incoming call is bridged to the H.323 connection so that both parties can talk to each other. As soon as the call is dropped on one end, the other call is dropped, too.

Outgoing calls from the IP network

If a network user who wants to make a phone call tries to enter the phone number in his H.323 application the call will never be connected since the application does not know how to connect to phone numbers. In this case a Gatekeeper can be defined in H.323 applications to which all outgoing calls are connected to and further processed by. In order to make outgoing calls, the server running TERAVoice needs to be configured as a gatekeeper in the H.323 application. Now TERAVoice will receive all outgoing calls and can connect them to the desired target. Upon successful connection the calls are being bridged to enable conversation.
Authentication

Every computer name or IP address on the IP network that should be allowed to use gateway functionality needs to have a Call Handler configured. If TERAVoice receives a call from the IP network from a computer that does not have a corresponding H.323 Call Handler the call is being rejected. This serves as a security measure in order to prevent unauthorized telephony connections.

Please note: This restriction cannot be disabled because it is also used by TERAVoice for licensing. The maximum number of allowed Call Handlers depends on the active license and disabling this restriction would allow an unlimited number of users to use H.323 gateway services. If you need a version that does not include this limitation, please contact sales@terasens.de on how to get an appropriate license.

IVR MODULE

Many requirements for IVR applications are so specific that a telephony application cannot provide the desired functionality out of the box. Things like database accesses, validating user input, playing back combined messages are so specific that some kind of programming needs to be involved. TERAVoice provides an API with programming interfaces that are easy to use and very flexible. Two ways for accessing the API are supported:

- Scripting
  Allows the writing of simple scripts in the user’s preferred scripting language

- ActiveX
  Allows the creation of ActiveX components in a COM-enabled programming language of the user’s choice

More information about developing IVR applications with programming samples can be found in the chapter Developing IVR Applications (p. 115).

MUSIC ON HOLD (MOH)

With PBX systems calls can be put on hold, for example if the caller is not intended to hear the assistant talking to another person, or calls are put on hold automatically if a call is about to be transferred to another station.

During Hold music or informational message can be played to the user. Most PBX systems allow feeding audio to be used during hold for all callers that are currently on hold from an external source.

Not all PBX systems allow feeding audio via a device connected to a phone line. TERAVoice can feed audio into the PBX via its Music on Hold Call Handler only with systems which provide this function. TERAVoice offers two modes of MOH feeding:

- PBX Initiated
  TERAVoice waits for the PBX to connect to the MOH Call Handler and starts playing audio until the PBX disconnects or TERAVoice is shut down
External Initiation
Upon start up TERAVoice connects on the specified devices and dials a configured number (if required). TERAVoice starts playing back the selected audio file until it is shut down. If the PBX closes the call, TERAVoice automatically tries to reconnect.

For more information about supported audio files please take a look at Supported Audio Formats (p. 59). For details on configuration and setup please read Setting up CallHandlers – Music on Hold (p. 101).

REMOTE CONTROL
Each Mailbox Call Handler can be configured to allow entering remote control mode by pressing ‘#’. Sometimes this behavior may be undesired and in other cases a central access point for accessing remote control for mailboxes may be preferred. For example if mailboxes are configured to answer calls after a certain delay, it is ineffective if the user needs to await the end of this delay in order to listen to his mailbox in remote control mode. For the more effective handling of these cases the Remote Control Call Handler can be set up.

Authentication
If a user calls the number associated with the Remote Control Call Handler the system basically does not know which mailbox the caller wants to gain remote control access to.

The standard procedure requires entering the ID of the desired mailbox followed by the user’s PIN number.

Another way of authenticating users is to allow authentication by Caller ID. For every Mailbox a list of Caller IDs can be configured that are allowed to enter remote control mode without entering their PIN number. For example this could be their PBX extension number, a mobile and a home phone number.

If the Caller ID is allowed to use remote control on a unique mailbox (i.e. the Caller ID is allowed for remote control on exactly one and not more than one mailboxes) the remote control mode can be entered without entering mailbox ID and PIN.

For more information on configuring the Remote Control Call Handler please refer to Setting up Call Handlers – Remote Control (p. 67).

The navigation in remote control mode is described in Using Remote Control from a telephone (p. 70).

TIME SCHEDULE
Sometimes the actions that have to be carried out on a call depend on factors like

- the Day of the week
- the Current Date
- the Current Time

The Time Schedule Call Handler is a method for handling those cases without programming.
**Actions**
The following actions can be configured to be carried out according to the selected schedule:

- **Transfer call**
  Transfers the call to another phone number

- **Go to menu**
  Hands over processing to the selected voice menu

- **Forward to Call Handler**
  Forwards the call to a different Call Handler

- **Hang up Call**
  Disconnects the call

For information about Call Transfer please read **Call Transfer Types** (p. 55).

**Unanswered Calls**
The Time Schedule Call Handler is the only Call Handler that can also operate on calls that have not yet been answered. This function is extremely useful if the time until the call should be taken also depends on the schedule. For example:

- **During business hours (like Mo-Fr from 0900h to 1700h):**
  Forward to voice mailbox "Day Mailbox" which answers calls after 30s and plays a message "We are currently unable to answer your call, please try again later…"

- **At all other times:**
  Forward to voice mailbox "Night Mailbox" which answers calls immediately and plays a message like "We are currently not available, please try again during our business hours…"

For instructions on how to configure Voice Menus please refer to **Setting up Call Handlers – Voice Menu** (p. 39).

**WAITING QUEUE**
Waiting Queues are used in most modern call centers in order to effectively distribute calls among agents and improve customer experience by optimizing waiting times and entertaining the callers until they get connected.

Waiting queues first take the call and play an announcement with some information. The Waiting Queue handler then determines the queue position by counting all other calls in the current queue that have not yet been connected. As soon as position one has been reached, the Call Handler starts trying to connect the call to one of the agents that are active on the queue. This is performed by calling all agents (the order depending on the selected cycle mode) and transferring the call as soon as a free agent has been reached. The call is then deleted from the queue.

**Cycle Modes**
There are several modes available by which the Call Handler tries to find a free agent:
- **Linear by list**
  The list of agents is cycled from the last agent a call was connected to until the end of the list and then starts from the beginning.

- **Linear by list (Reset start)**
  The list of agents is cycled from the start of the list to the end of the list and then starts from the beginning. This mode will prefer agents from the beginning of the list, so that in times with less activity, agents from the end of the list do not get many calls and thus can be kept as some kind of "backup" for times of high activity.

- **By order of transferred calls**
  This mode will always try the agent first that did not receive a call for the longest time and can be considered to be the mode with the most equal distribution of calls to agents.

Please note that a decision about which agent to call first always depends on the time at which a call was last transferred to an agent. The system does not know how long an agent's call actually takes and therefore cannot take this into account when distributing calls. Nevertheless over some period of time the distribution of calls among agents should be approximately equal using either the first or the third cycle mode.

**Queue Full Condition**

In order to improve customer satisfaction and keep waiting times from getting too long, a maximum number of queue slots can be configured. If a caller gets connected to a queue which is full (i.e. all slots occupied) a message can be played back and an action can be selected to be carried out. The types of these actions are:

- **Transfer call**
  Transfers the call to another phone number.

- **Forward to Voice Mailbox**
  Forwards the call to the selected voice mailbox.

- **Forward to Call Handler**
  Forwards the call to a different Call Handler.

- **Hang up Call**
  Disconnects the call.

**Announcements during wait**

During the waiting time a call is usually put on hold, so that the caller can listen to the Music on Hold that is played by the PBX. If desired an announcement like "Please wait, we will try to connect you as soon as possible" can be played at a selectable interval. In order to play back that message the call is fetched back from hold state before playing and put back on hold afterwards.

**Announcement of position**

Optionally a message can be played each time a caller's position in the queue changes; this message informs the user about his current queue position, e.g. "You are currently at position number x in the waiting queue."

This type of information can greatly improve customer satisfaction since the user can estimate the remaining time until being connected to an agent from these announcements.
Call Handler Assignment

In the previous chapters a lot of information on Call Handlers and their functions were covered. After the creation of a Call Handlers, how it be determined what Call Handler is used on a specific call?

TERAVoice allows the use of any information it can get from a call to assign Call Handlers to that call. The following criteria are available:

- By Device (p. 46)
- By MSN/Extension (p. 46)
- By Redirector (p. 47)
- By Inband Signaling (p. 48)

For a description of each item please refer to the corresponding topic. Which of those options are useful depend on the network configuration (analog/digital, PBX/no PBX). Before creating Call Handler assignments you should enable the options that you intend to use for assignment. For more information how to do this please read System Configuration – Call Handler Assignment (p. 95).

Example

You can apply one or more assignment entries for each Call Handler. For example you are using Device, MSN and Redirector information for assignment. You have a modem and an ISDN device and want to setup a mailbox for the user with extension 235. The ISDN device is connected to the PBX and is operating on extensions 170-190.

First you want to configure a corporate mailbox on extension 170. The PBX offers all incoming calls at extension 0 and 170. The assignment entry for this mailbox will look like:


Now you want to configure a voice mailbox for user 235. User 235 sets a call diversion to "his" diversion extension 185. The assignment entry for this mailbox will look like:


Alternatively, to avoid using an extension for each mailbox redirection, if the PBX transmits redirector information, all users can divert their call to the same extension 180. The assignment entry for this case will look like:


Matching Assignment entries

On incoming calls TERAVoice will always try to find a match for the current parameters. If no match is found, TERAVoice will use the Default Call Handler.
Default Call Handler
The default Call Handler can be set by right-clicking on a Call Handler and selecting "Set as default". Only one default Call Handler can exist. The default Call Handler is indicated by the following icon: ▶.

Instructions on how to configure assignment entries can be found in Setting up Call Handlers – Basic Options (p. 95).

BY DEVICE

Call assignment via Device is very simple. The Call Handler is using the name of the device that reports an incoming call for assignment of the matching Call Handler.

When should I use assignment by device?

- If you are using only one device (this means only one entry selected in the device list) then you do not need to use Call Handler assignment by device.
- If you have several selected entries in the device list but you do not care on what device a call is coming in you do not need to use assignment by device either.
- This option is useful if you have several devices that are active on different phone numbers or you are using a mixed configuration of devices (like an ISDN board and a voice modem) that serve different purposes.

Device Groups

In Addition to devices you can also select device groups for assignment. The reasons for device grouping are laid out in Device Groups (p. 37).

On how to enable Call Handler assignment by device please read System.Configuration – Call Handler Assignment (p. 78).

Instructions on how to configure assignment entries can be found in Setting up Call Handlers – Basic Options (p. 95).

BY MSN/EXTENSION

Especially in digital telephony networks but maybe also with PBX specific telephony boards one line connected to a telephony device can not only act on more than one phone number (MSN or station/extension) but also report which number was called. This number is generally referred to as Called ID.

When should I use assignment by MSN/Station?

- You should always use this kind of assignment if at least one of your telephony devices supports reporting Called ID information.
- The main benefit is that you can implement different service types with just a single telephony device acting differently depending on the number that was called

Example: You have an ISDN board behind a PBX which is set to MSNs 15, 16 and 17. Now you can configure a corporate mailbox for MSN 15 and your call center waiting queue for MSN 16. The central number for all user mailboxes is MSN 17. (The mailboxes are assigned by the Redirector ID).
If your system does not support reporting of Redirector IDs then you should configure one MSN for each mailbox: Users’ extensions are from 20-39 you configure MSNs 80-99 for your telephony device. Each user redirects calls to his mailbox’s MSN.

On how to enable Call Handler assignment by MSN/Station please read System Configuration – Call Handler Assignment (p. 78).

Instructions on how to configure assignment entries can be found in Setting up Call Handlers – Basic Options (p. 95).

In a PBX system or in a public ISDN network every telephone port or line can have one or more extension numbers.

**BY REDIRECTOR (REDIRECTING ID)**

Usually this information is only available in ISDN networks. The Redirector ID designates the number from which the call was redirected to the current target. In detail: when a user A calls B and B has set a call diversion to C the call from A will be redirected from A to C. C will receive the phone number of A as the Caller ID. If the network supports that feature, C will receive the number of B as the Redirector which is called the Redirector ID.

**When should I use assignment by Redirector?**

The Redirector information is normally only useful for voicemail applications in order to determine which mailbox to chose. Using Redirector information is just easier than configuring one separate mailbox phone number for each mailbox. If your telephony device or network does not provide Redirector information then you do not need to activate this feature.

---

**Example:** You have configured a telephony device on MSN/extension 15. All users redirect their phone to this number (or you put this number in an appropriate PBX ringing group) when they want to activate their mailbox. Your assignment entries will look like this for all user mailboxes:


....

You can create a Remote Control Call Handler with the following assignment:


So each time a user calls the mailbox number (15) directly he will get into remote control mode. If Auto-Login by Caller ID is activated he will listen to new messages right after connecting.

---

On how to enable Call Handler assignment by Redirector ID please read System Configuration – Call Handler Assignment (p. 78).

Instructions on how to configure assignment entries can be found in Setting up Call Handlers – Basic Options (p. 95)..
BY INBAND SIGNALING (DTMF DETECTION)

If you want to connect your voice mail application to an analog PBX, the PBX needs some way of signaling to the application which mailbox a call is targeted for. Currently we know of two methods that are commonly used for this purpose:

- Direct Connection from PBX to the voicemail server via a serial cable connection (RS232)
- Inband Signaling

**Note:** TERAVoice currently only supports Inband Signaling.

**How does Inband Signaling work?**

With Inband Signaling the PBX sends a certain sequence of DTMF digits right after the call has been taken by the PBX system. After the PBX has sent the information it connects the call to the voicemail server.

The sequence of digits that is sent differs from PBX to PBX. Usually the following information is transmitted with this sequence:

- Caller ID
- Called ID (extension number of mailbox owner)
- Type of call: external direct, external redirected, internal direct, internal redirected

What TERAVoice needs to do is to somehow extract this information from the digit sequence. This is done by using pattern matching functions. For each type of call a pattern must be entered containing actual digits [0-9] that need to be contained, a placeholder [%] that contains the Called ID and a special character [X] to ignore a digit. The Caller ID is currently not processed by TERAVoice.

**Example:** The following example is based on the Inband Signaling patterns for a SIEMENS HiCom or HiPath with a 2 digit internal numbering plan.

- Direct internal: ***1***%*1
- Direct external: ***202222*4
- Redirected internal: ***3***%XX*2 (use XXX instead of XX for 3 internal digits)
- Redirected external: ***302222%*4

The number that is received from the % placeholder is the parameter that is used in the assignment entries of type DTMF. The length of the number represented by the % sign depends on the configured internal number digit length.

Depending on the type of call TERAVoice automatically either starts in mailbox mode or remote control mode.

A step-by-step description of how to configure Inband Signaling with your PBX can be found in the topic *How to configure Inband Signaling* (p. 71).
On how to enable Call Handler assignment by Inband Signaling please read *System Configuration – Call Handler Assignment* (p. 78).

Instructions on how to configure assignment entries can be found in *Setting up Call Handlers – Basic Options* (p. 95).
Network Configurations

Several types of system configurations exist, each with its different requirements and possibilities regarding implementation of certain features. In order to get information on your type of configuration, please choose your type of network first:

- Analog Telephony Network (POTS)
- Digital Telephony Network (ISDN)

Of course there are a lot of scenarios where mixed configurations are present, like a PBX that offers analog and digital ports or configurations where the internal network type is different from the external type. In these cases information from both topics might apply.

ANALOG NETWORKS

Analog Networks

POTS (plain old telephony system) is referring to the analog type of telephony networks that have been present for the last decades and still connect the majority of telephony subscribers in most countries. Calls are being made by transmitting either audio pulses or DTMF (dual tone multi frequency) tones to the provider’s switch which then tries to further connect the call.

Without PBX

If you have a configuration with an analog connection and you are not using a PBX you are probably running a small office where you want to implement some kind of voice mailbox or automatic attendant system.

You are probably running a voice modem or an analog telephony card with only one or few channels.

Please note that this kind of configuration is very limited because a lot of features, like Call Transfer or Call handler assignment methods except "By Device", are not available.

If you are using more than one device (line) you can use Call Handler Assignment - By Device (p. 46) in order to offer a different service on each line.

The most common configuration for this setup is a Voice Mailbox-Call Handler that is set as a default Call Handler (or assigned to a specific device if using more than one device), where the Voice Mailbox or other Call Handler is set to answer after a certain time and the telephone is working parallel to the telephony device.
Please note that the figure only shows an example configuration. Due to the many different network standards, connection types and equipment specifications that are available, your actual configuration may differ.

Procedure where A calls B. B creates a consultation call to C while holding A. B then connects/transfer A to C.

**With PBX**

If you are using an analog PBX or alternatively a PBX that offers analog internal ports you will of course need an analog telephony board or a voice modem to connect to your PBX.

Calls can be assigned to Call Handlers by the following assignment methods:

- **By Device (p. 46)**
  
  If you are using several lines that are connected to your PBX, each of these lines would appear as a device in the device list (depending on your telephony board’s TAPI driver). To offer a different service on certain lines/extensions you can use "Call Handler Assignment By Device". You can also use Device Groups in order to simplify configurations with a large number of devices.

- **By Inband Signaling (p. 48)**
  
  For voicemail applications you can use Inband Signaling to inform TERAVoice which Call Handler (Mailbox) to choose for an incoming call if your PBX supports Inband Signaling.

Advanced features like Call Transfer or Call Hold require the support of your telephony board. TERAVoice does not allow defining Hook/Flash sequences that your PBX requires in order to execute the desired feature. These sequences need to be defined with your telephony board; TERAVoice then calls the required high-level function on the TAPI driver of your telephony board.
Figure - Analog network with PBX

' Please note that the figure only shows an example configuration. Due to the many different network standards, connection types and equipment specifications that are available, your actual configuration may differ.

Procedure where A calls B. B creates a consultation call to C while holding A. B then connects/transfer A to C.

DIGITAL NETWORKS

ISDN is the world-wide standard for digital telephony networks. ISDN offers a lot of advantages over the standard analog telephony system, such as the assignment of several numbers to a single line or the transmission of advanced call information, such as details about redirected or diverted calls.

There are also PBX specific standards for digital call transmission. These PBX's require certain PBX specific telephony boards. The network features that these boards offer may differ from the common ISDN features. Please consult your PBX documentation and the documentation about the TAPI features that you telephony board offers for more details.

Without PBX

Usually ISDN networks provide a bus interface where telephones, ISDN boards and other telephony equipment can be plugged into. ISDN offers the ability of assigning several phone numbers called MSN’s (Multi Subscriber Number) to an ISDN installation. This provides the ability for all connected devices to act on different phone numbers and from a telephony server’s perspective to offer different services on a single connection or device.

In this configuration type the following methods for Call Handler Assignment are usually used:

- By Device
- By MSN

The set of features that is available with this configuration depends on the supported features of the public network you are connected to. For example the network in Germany (Deutsche Telekom) supports 3-Party Conferences (by default) and even Call Transfer (separate subscription is necessary for this feature), while other network providers may only support a smaller set of features.
With PBX

This type of configuration usually provides the widest range of features. Depending on the type of interface that your PBX supports for the connection of telephony boards you need to select the appropriate telephony hardware for use with TERAVoice. As always the set of features that can be used with TERAVoice depends on the features that your PBX supports in combination with those of the telephony board and the features that the telephony board’s TAPI driver offers.

For Call Handler assignment the following methods can be used:

- By Device (p. 46)
- By MSN/Extension (p. 46)
- By Redirector (p. 47)
MESSAGE WAITING INDICATION (MWI)

Some PBX systems offer a feature called Message Waiting Indication (MWI) which is used to signal to the user that there is a new voice mail or text message available. Usually this condition is signaled by a light on the user’s telephone. Some telephones also provide a button near the indicator light that allows displaying the new text messages or dials into the voice mail system in order to listen to new voice messages.

PBX systems with MWI features offer an interface that allows external voice mail systems to turn on and off the MWI lights. The most common method (and the only method that TERAVoice supports) is to have the voice mail system dial a certain number in order to control the MWI lights.

Example for SIEMENS HiCom/HiPath:

Turn on MWI light: ‘7568%0’

Turn off MWI light: ‘7668%’

Where ‘%’ is the extension number of the station that is to be controlled.

Please note: For SIEMENS systems this method only works if the station that is calling the code is configured as 'Voice Mail'.

Please consult the documentation of your PBX system for more information on how to set up MWI with other models.

In a PBX system or in a public ISDN network every telephone port or line can have one or more extension numbers.

TEXT-TO-SPEECH

In addition to predefined audio files that can be used for playback in TERAVoice you can also directly enter speech that is generated by a TTS (Text-To-Speech) engine. More information on how you can specify text or audio files to be used can be found in chapter Setting Up Call Handlers (p. 95).

Currently TERAVoice can only use one TTS Voice which can be selected in the basic configuration options. More information on how to select a TTS Engine and voice can be found in topic Server Options (p. 91).

TTS Engines

TERAVoice uses the Microsoft Speech API Version 5.1 (SAPI), which is automatically installed with TERAVoice. Currently TERAVoice does not include a TTS engine though. There are some TTS engines installed that are part of the Microsoft Speech API but the quality of those engines is not very impressive. There are a lot of speech engines that you can license from their manufacturers or just download for evaluation first. Microsoft also offers some engines that are automatically installed with certain products.
CALL TRANSFER TYPES

Call transfer is the process of handing over a call to another station. TERAVoice can use several methods for achieving this functionality.

Call Transfer Example: A calls B and talks to B. B calls C while holding the call to A. B transfers the call. As a result A is connected to C; B is not connected to anyone.

This topic describes all available methods for Call Transfer. For information about configuring the transfer mode that should be used for your configuration please read Server Options (p. 91).

Native Call Transfer

Usually call transfer is a feature that is offered by PBX systems while a few public ISDN networks also offer this feature directly (the German network of Deutsche Telekom offers this feature on request).

For ISDN hardware call transfer is a network feature that is invoked by sending certain data in the D-channel protocol. In order to be able to use this call transfer you need to be connected to a PBX or public network that supports call transfer and your telephony hardware needs to support this feature.

For analog devices the device must allow the programming of the Hook/Flash sequences that are required by the PBX in order to invoke call transfer.

TERAVoice offers several modes for native call transfer. If you are unsure which mode to choose it is best to test different modes, and then choose a mode, that is working. Not every mode will work with every configuration.

Wait Until Connect, Then Transfer

This is the most common transfer mode and should always be tried first. TERAVoice tries to connect to the station that the call is to be transferred to (C-party) and waits until this consultation call gets answered; the transfer is then completed.

Immediate Transfer If not Busy

This mode is a special kind of transfer where the transfer is completed right after the consultation call is ringing at the C-party. This method is the most unlikely to work among all transfer methods. Some PBX’s support this method, though. If you are using an ISDN board with a CAPI interface you can try this method if your PBX supports it.

Please note that it depends on your PBX what happens to the call if the C-party doesn’t answer the call. Usually the call is handed over to a default ringing group or is signaled again at the station that initiated the transfer.
**TAPI Blind Transfer**
This is a special transfer method offered by TAPI. It is very similar to "Wait Until Connect, Then Transfer" but the consultation call is made by the TAPI driver rather than by TERAVoice. The disadvantage here is that TERAVoice has no control over the consultation call. If the C-party doesn't answer the call TERAVoice will never regain control of the call. The A-party can only hang up the call in this case.

**Warning:** The only reason why this method was included is that there is at least one TAPI driver we know of that supports only Blind Transfer but not the normal transfer method. Please do not use this method unless the "Wait Until Connect, Then Transfer" method fails and only "Blind Transfer" works.
You also should use this transfer type only in single-line configurations. An ongoing BlindTransfer will block everything else. (This is a TAPI problem, not a TERAVoice issue)

**Simulated Transfer**
There are quite a lot of configurations where the native Call Transfer functions won't work. TERAVoice includes two additional options to simulate Call Transfer functionality.

**3-Party Conference**
Conference is a feature that is supported by some public ISDN networks and by many digital PBX systems. While PBX systems usually offer Call Transfer functionality most public networks do not (at least not by default), while some offer at least the 3-Party Conference feature. In this mode TERAVoice simply creates a conference instead of initializing Call Transfer and silences the party initially called during the conference.

The disadvantage of this method compared to the native Call Transfer methods is that one line stays connected during the conference and is not available for handling other calls. Therefore this mode should only be used if the native modes fail.

**Call Recording:** One additional reason exists for using the 3-Party Conference mode. If you want to record calls that are transferred to call-agents via the Waiting Queue Call Handler (p. 105) and you want to record the conversations, you need to use this mode. TERAVoice creates the conference after the call is connected to the C-party and starts recording until A or C disconnects.

**Bridged Transfer**
If all other methods for Call Transfer should fail you could still use the Bridged Transfer mode. When doing Bridged Transfer TERAVoice uses an additional line to dial out to the C-party. Upon connection TERAVoice starts bridging the calls by copying audio data from one call to the other and vice versa.

One drawback with this method is that 2 lines are occupied while doing Bridged Transfer. It is best to try all other methods before using bridged transfer and maybe purchasing hardware that supports native transfer methods should be considered instead. The amount of lines needed for TERAVoice licensing and for the telephony hardware supporting the required number of lines can add up to more than the cost for different hardware.

**Note:** Bridged Transfer requires telephony hardware that supports full-duplex mode. Usually all telephony boards except for voice modems (using the Windows Unimodem/5 Driver) support this feature!
Simulated Hold

For all other transfer modes the first call is put on hold while the consultation call is being made. While the first call is on hold a message or sound is played back to the caller (Music On Hold), which is generated by the PBX or the public network. For simulated transfer you can specify an audio file that is to be played if it is not possible to put the call on hold. You can also choose to always play the simulated hold audio file instead of trying to put the call on hold.

SENDING SMS AND PAGER NOTIFICATIONS

In order to inform user about new messages in their voice mailboxes, TERAVoice supports sending notification messages to pagers and mobile phones.

Currently there are two major standards available which are used by paging service and mobile phone network providers in order to send those messages:

- TAP (Telelocator Allocator Protocol)
- UCP (Universal Computer Protocol)

The submission of SMS/pager messages is accomplished by dialing into the provider's remote system via analog modem (in rare cases also via ISDN). The data according to the required protocol (TAP or UCP) is then transmitted and the connection is closed.

SMS/Paging Services

TERAVoice comes with a large number of predefined paging services and SMS gateway services. Before using and configuring a service you should always try to send a test message using the test tool found on the Diagnostic Tab of the SMS/pager configuration dialog.

For more information on how to configure SMS/paging please refer to SMS/Pager Configuration (p. 85).

More tips for configuring SMS and pager settings can be found on:

   http://www.terasens.de/fwlink.asp?q=35042

Services and Routing

TERAVoice supports the definition of multiple services that can be configured for sending SMS/pager messages. Every message that is to be sent requires a target phone/pager number that the message is intended to be sent to. Depending on this target number you can set up a routing strategy in TERAVoice.

**Example:** You have installed two services, Service 1 and Service 2. All messages to numbers that start with 0151 should be sent via Service 1 and all numbers to 0152... should be sent via Service 2.

You can also define a default service that should be used if no explicit routing entry is present.
LANGUAGES SUPPORT
Language support in TERAVoice can be divided into two different areas: Software User Interface Languages and Voice Prompt Language.

Software User Interface Language
Currently TERAVoice supports the following languages for the Software UI:

- English
- German

When you start the installation process a dialog is displayed which lets you choose between those languages. The selection made in this dialog affects the following items:

- Language of the setup program
- Language of the Shortcuts in Start-Menu
- Language of the help files
- Language of the documentation
- The Default Language for voice prompts

The language of the administration console and the language of the logging information that is generated by the TERAVoice server service depend on the language of the operating system. If the operating system is of a language that is not supported by TERAVoice then English will be used.

Voice Prompt Language
For every type of application you create you can of course record voice prompts in any language that is desired. TERAVoice comes with a set of predefined voice prompts, though, which are currently provided in the following languages:

- English
- German

These predefined voice prompts are used for the following purposes:

- Default Greeting Message (1 audio file)
  This is used for mailboxes for which no a specific greeting message has been defined

- Default Good-Bye Message (1 audio file)
  This is used for mailboxes for which no specific good-bye message has been defined

- Remote control menus (43 audio files)
  These are used to inform the user about results and options in Remote Control mode

The default greeting and good-bye messages can be easily replaced by specifying your own audio files as described in the topic Server Options (p. 91).
If you want to localize the remote control menu voice prompts please visit our web site:

http://www.terasens.de/fwlink.asp?q=35041

**SUPPORTED AUDIO FORMATS**

TERAVoice uses Windows DirectShow interfaces for playback of audio files. This is the most flexible method because a broad range of audio formats and audio codecs is supported.

As a general rule TERAVoice can play all audio formats that you can play back using the Windows Media Player on your system. You can even use video files if you want to use a video’s audio track.

While this kind of audio support is a nice feature you should not forget the amount of system resources that is necessary to convert audio data from the selected audio file to the format that the telephony hardware requires. On modern systems with just a few lines this is of minor importance, but if you are creating a professional telephony solution operating on more than a few lines you should use only audio files that are of the same format that your telephony hardware supports in order to avoid sample rate and bit-rate conversions, thus saving system resources.

Most telephony hardware is using the following audio format: **8kHz - 16bit - Mono**

Note: The easiest way to find out the audio format your telephony board supports is to create a mailbox and record a message. Then you can open the recorded file and examine the audio format.

Recorded audio files are always saved in WAV-format. TERAVoice currently does not allow realtime encoding into other formats (such as MP3) in order to save system resources.

**SECURITY CONSIDERATIONS**

TERAVoice offers an interface for remote administration and mailbox access for users via the TERAVoice Client Application. Both of these mechanisms use the same method for gaining access to TERAVoice data. This is done via a hidden network share.

Important: If you want to allow users access their mailbox information via the Windows Client Tool it is highly recommended to install TERAVoice on an **NTFS** partition. FAT or FAT323 partitions do not allow storing security information at a file and folder level. This security information is crucial for setting security for users' mailboxes in TERAVoice.

During installation the network share is automatically created by the setup program. The shared folder is called ‘TVShare’ and is located in the root installation folder of TERAVoice. By default this is the following path:

```
C:\Program Files\TERAVoice Server 2004\TVShare
```
This folder is shared as 'TVShare_0$' where the '$' sign in the name causes this share to be hidden when browsing the network. If you do not need remote administration or users' mailbox access via the Client Tool you can safely delete this share (please never delete the folder itself).

**Note:** Please make sure that the TERAVoice service itself is allowed to access all files. By default the TERAVoice service is running under the LocalSystem account but for some reason you might have changed this (e.g. for debugging). Please make sure that the account the service is running under is granted permissions for full access to all files and folders!

To properly secure your system permissions need to be set in the following way:

**Remote Administration files**
All Files in the TVShare folder should have only administrator permissions. These files are used for accessing TERAVoice configuration, log files and status information. Normal users should not have permission to read these files. These files are named:

- status.srf
- config.cfg

**Mailbox Folders**
The folder MBX contains all mailbox folders. The name of each folder corresponds to the ID of the user’s mailbox Call Handler. Every user should have only permission to read the mailbox folder he should be allowed to access.

**Tip:** The property sheet of the mailbox Call Handler automatically displays a security page as the rightmost tab that lets you set permissions for all mailbox folders in a convenient way.
3 Using TERAVoice

While the previous chapter, Basic Concepts (p. 27), explained the basic technology, terminology and features the following chapter is going to focus on how to set up and configure TERAVoice in order to achieve the desired functionality. Information about advanced IVR functions applications that need implementation by programming or scripts is given in the next chapter, Developing IVR Applications (p. 115).

The first topic in this chapter, Common Tasks (p. 62), gives step by step instructions on how to configure TERAVoice for certain basic types of functionality.

The topic Administration (p. 75) describes all configuration tasks and dialogs that are available in the TERAVoice administration console.

Monitoring System Activity (p. 107) provides help on how to monitor present and past activity on your TERAVoice server.

The TERAVoice Client Application is described in the last topic of this chapter.
Common Tasks

HOW TO SET UP TELEPHONY DEVICES

The first step in setting up your telephony devices is to install your telephony hardware and the TAPI drivers or CAPI drivers that come with your hardware. If there are special configuration tasks necessary for your hardware please complete these tasks according to your device’s documentation.

Then reboot your system before continuing.

Test Your Hardware

Before you try to use your hardware with TERAVoice you should test whether your hardware is installed and working properly. A good method for testing the TAPI driver of your hardware is the Windows Phone Dialer (Start ⇒ Programs ⇒ Accessories ⇒ Communications ⇒ Phone Dialer or Start ⇒ Run ⇒ Dialer.exe ⇒ OK).

Open the Phone Dialer, select Edit ⇒ Options and choose the device you want to test in the combo box for "Phone Calls". Select "Phone" as the preferred line for calling. Close the dialog.

**Note:** If you are using a CAPI based ISDN board, your device will show up as **ISDN Interface x (Channel y)** in the device list. You need to reboot after installing your CAPI drivers in order to have them appear in this list.

Click on the Dial button, select "Phone Call" and enter the number you want to dial. If you are behind a PBX please keep in mind that you might need to first dial a certain digit in order to get an external line. You also might simply try calling an internal extension.

Click on Place Call and watch the call progress. If you can successfully connect the call everything should be fine and the device should be ready for use with TERAVoice.

If you have a sound card with speakers and microphone installed you should be able to talk to the other party. If this does not work although you have selected your sound card's devices on the Audio/Video tab of the options dialog and you have audio problems with TERAVoice as well, there is a problem with your telephony hardware’s driver or your TAPI driver does not support audio (there are TAPI drivers for telephones, PBX’s and other devices available that allow only dialing via TAPI but do not support audio transmission. This type of application is called Assisted Telephony).

**Note:** Although it is helpful for several reasons to have a sound card installed on your TERAVoice machine this is not required. TERAVoice will run just fine without a sound card installed!

If you have any problems completing this test please refer to chapter Check Your Telephony Hardware (p. 148) or read the documentation that comes with your hardware.

Configure Devices

If you have successfully tested your device you can continue to configure your device for use with TERAVoice. Make certain that you have shut down the Phone Dialer after testing and that it does not show up in the notification area of the system tray.
Open the TERAVoice Administration Console, expand the Server Configuration node and right-click Telephony Hardware; then select Properties.

Now check all devices that you want to use with TERAVoice on the list on the first tab. When you click on Device Capabilities you will get a dialog with details about the features that your device supports. A description of those features can be found in the topic Device Features (p. 34).

**CAPI Devices**

If you are using ISDN boards with a CAPI driver your device will show up in the device list as ISDN Interface x (Channel y). In most cases no further configuration will be necessary. To configure the CAPI driver settings select one of the devices and click on Configure Device. For details about CAPI configuration please refer to the topic CAPI Configuration (p. 80).

**Device Settings**

The second tab Device Settings lets you specify some advanced options for each device you have selected. The settings on this dialog are device dependent, so you need to repeat all settings for each device.

**Modify Phone Numbers**

If your device is connected to a PBX the PBX might signal external calls with one or more additional leading digits (like '0' or '9') that need to be dialed to get an outside line. If you want to remove these digits you can specify this here. If you do not want to remove the specified digit for internal calls you should select Only if equal to and enter the digit that should be removed.

**Audio Delay**

If the voice prompts are not audible from their beginnings, you can select a delay for the device the problem occurs on. The delayed audio setting is device dependent but the actual delay time is a global setting which is not device dependent.

Note: Usually this setting is only required for voice modems with Unimodem/5 driver!

**HOW TO SET UP VOICE MAIL**

Setting up TERAVoice as a voice mail system requires several steps depending on your network configuration and the desired features. Before following the instructions in this topic you should first configure TERAVoice to use your telephony hardware as described in the previous topic How to Set up Telephony Devices (p. 62).

**Call Handler Assignment**

The most important thing when setting up TERAVoice as a voice mail system is to choose the right assignment method which is responsible for choosing the right mailbox for an incoming call. General information about this issue can be found under Call Handler Assignment (p. 45). Assignment methods that are available for certain network configurations are described in the chapter Network Configurations (p. 50).

The following suggestions cover the most common scenarios of telephony and PBX systems and configurations but there might be cases - especially with mixed configurations - where other combinations of settings might be required. The assignment method you should choose depends on the configuration you use:
Analog Network without PBX

The only option here is to assign a different mailbox Call Handler to each line/device if you are using more than one line/device and have each line connected parallel to each telephone you want to provide a voice mailbox for.

- Call Handler Assignment By Device (p. 46) should be used

**Note:** If you have a telephony board that supports more than one line it is required that your board's driver installs a TAPI device for each line in order to use assignment by device. Alternatively it might be possible that your board's driver lets you assign a phone number for each line it supports which is signaled as CalledID. In this special case you can choose Call Handler Assignment by MSN/Extension (CalledID) (p. 46).

PBX with analog internal ports

For this configuration the same information applies for the previous configuration (without PBX).

Additionally some PBXs support transmitting the extension the call was targeted at via Inband Signaling. In case of Inband Signaling the telephony board is not connected parallel to the telephones but rather works on one or more analog ports of the PBX. The PBX then redirects calls to the mailbox after a certain time or immediately if the user has activated his mailbox in this way.

So you should use:

- Call Handler Assignment By Device (or see previous note) if your PBX does not support inband signaling, or
- Call Handler Assignment By Inband Signaling (p. 48) if your PBX does support inband signaling

More details on configuring Inband Signaling can be found under How To Configure Inband Signaling (p. 71).

ISDN networks without PBX

In this configuration the telephony hardware is usually connected to the same line as the users' telephones. As opposed to analog networks, multiple numbers called MSN's can be assigned to a line. Whenever a call is signaled on the line the MSN that the call is targeted at is transmitted and only those devices that are configured to act on this MSN try to answer the call. Thus, several different mailboxes can be configured (one for each available MSN) in this case.

In addition to that of course the use of more than one device each connected to a different line is possible and the use of assignment by device as well.

So you should use:

- Call Handler Assignment By Device (p. 46) if you are using more than one device, or
- Call Handler Assignment By MSN/Extension (p. 46) if you have more than one MSN number available on a line, or
- Both methods if both conditions apply
Using TERAVoice - Common Tasks

Note: If your public ISDN network transmits redirector information you could also use Call Handler Assignment By Redirector. The advantage here is that you can supply voice mailboxes for external subscribers i.e. subscribers that are not connected to your local line. Those subscribers would set up a call diversion to the phone number TERAVoice is working on to redirect calls to TERAVoice. The drawbacks here are first that the phone cost is higher because you might need to pay for the duration of the redirected call and second that not all public networks support transmitting the redirecting party’s number (while it is more common that a flag is transmitted indicating that the call was redirected but not by which party).

PBX with digital internal ports
One of the advantages of digital connections is the ability to assign more than one phone number to a single line. As opposed to the inband signaling method the PBX is configured to assign one number to the line (or group of lines) that is connected to TERAVoice for each mailbox. Using a certain assignment scheme can help simplifying configuration.

Example: Internal extension numbers are from 20 to 49. You can then use a simple rule (e.g. plus 50) to the lines that are connected to the TERAVoice server. As a result the user with extension 25 would redirect calls to his mailbox number: 75).

For systems that support redirector information the configuration is even easier by using Call Handler Assignment by Redirector. All users redirect their calls to a single phone number TERAVoice is working on. When a redirected call gets signaled the number of the redirecting extension is transmitted as the RedirectorID.

So you should use:

- **Call Handler Assignment By Redirector** (p. 47) if your PBX and telephony hardware support it, or
- **Call Handler Assignment By MSN/Extension** (p. 46) if redirector information is not transmitted and optionally
- **Call Handler Assignment By Device** (p. 46) if you are using several devices for different purposes

Information on how to activate or deactivate Call Handler Assignment methods can be found under **Server Options** (p. 91).

Voicemail Settings

SMTP Server
As a next step you need to configure several options for your voicemail system if you want to have TERAVoice send voice messages via e-mail. The first is to specify an SMTP server that should be used to send mail. You need to specify the DNS name or IP address of the SMTP server to use, the name and the e-mail address that should appear as the sender of voice mail messages and optionally logon information if the server requires authentication. More information can be found under **Server Options** (p. 91).

Miscellaneous Settings
Several default settings can be additionally configured like the default time before taking calls, the default maximum recording time, default greeting and good-bye messages and the default voice prompt language for remote control mode.

All these settings can be overridden by the individual settings of each mailbox.
Notifications

Now the notification options you want to use to inform users about new messages in their mailboxes should be configured.

Email Notification

There are two types of notifications TERAVoice can send via email: Voice Messages and Call Notifications. The latter type of notification is sent every time a call is received while the other is only sent if the caller has left a voice message. You can define which types of email notifications should be allowed to be sent and you can edit the template of the message that is sent. Which notifications are enabled for a specific mailbox can be configured in each mailbox’s settings.

SMS And Pager Notification

Notifications can also be sent to mobile devices via SMS (in GSM networks) or pager networks. Details about configuring SMS/Pager notification can be found under SMS/Pager Configuration (p. 86), for basic information about this type of notification please read Sending SMS and Pager Notifications (p. 57).

PBX Notification

This type of notification allows turning on and off the MWI (Message Waiting Indication) light on PBX telephones that support this feature. For details about how to configure this notification type please refer to Notifications (p. 66).

Setting up Mailboxes

Now the mailboxes you require can be set up and configured. Expand the Call Handlers node in the administration console, right-click on the Mailboxes node and select New Mailbox. Enter a name for your new mailbox and select OK. You newly created mailbox should now appear in the list.

Default Call Handler

TERAVoice needs to have a default Call Handler defined which is used any time a call can not be assigned a Call Handler according to the configured assignment rules. If this is the first mailbox you create (any Call Handler can be the default Call Handler, though), right-click the new entry and select Set as default.

Now right click the mailbox and select Properties.

General Options

On the first tab you can set name and description and the time TERAVoice should take before answering a call. Depending on the type of assignment and your network configuration you might want to set a certain delay (usually if the call gets signaled on a telephone and at TERAVoice simultaneously) or without a delay (usually if the mailbox redirects the call to TERAVoice after a certain amount of time).

Assignment Rules

If you are creating a mailbox that is not the default mailbox (i.e. default Call Handler) you need to set assignment rules depending on the Call Handler Assignment (p. 45) methods you have selected. You can enter more than one assignment rule for each Call Handler. More information about configuring these assignment rules can be found under Setting up Call Handlers - Basic Options (p. 95).

Greeting Messages

Every mailbox can have one or more greeting messages. By default only one message is enabled. The message can be either an audio file or a text that is played back via text-to-speech. If you do not specify a message for a mailbox then the default greeting message is played.

If required you can activate scheduled greetings for a mailbox. In this mode you can specify any number of messages that are played depending on date criteria like Weekdays, date ranges or time ranges.
You can also specify (even for each schedule) if recording a message should be allowed.

Notifications
You can specify for each mailbox which types of notifications should be activated and to which email addresses or phone numbers the notifications should be sent.

Remote Control
For each mailbox you can allow or disallow remote control. You can set a PIN code that is required for remote control access and you can also specify a list of CallerID’s that are allowed for remote control access without entering a PIN code.

Example: With CallerID authentication the user does not need to enter his mailbox PIN in order to enter remote control mode, e.g. if he calls from his mobile phone.

Options
Details on several other options that can be set can be found in Setting Up Call Handlers - Mailbox (p. 96).

HOW TO SET UP VOICE OVER IP BRIDGING (H.323)
There are a few steps that need to be taken in order to activate H.323 bridging with TERAVoice. First it is important that there are three types of calls that involve bridging to H.323 devices:

- Inbound calls that are bridged to a H.323 using the regular call transfer functions from any Call Handler that offers call transfer functions
- Inbound calls that have a fixed assignment to an internal H.323 device that is set by an H.323 User Call Handler (p. 99)
- Outbound calls - calls originating from an H.323 device that belongs to an H.323 Call Handler

The first type does not rely on an H.323 Call Handler. A call can be transferred to any IP address or host name with the regular call transfer functions.

The latter two involve a configured H.323 Call Handler. You need to create an H.323 Call Handler for every user/machine/device that you want to permit using H.323 gateway functionality.

Steps to Activate

Enable outbound H.323
First you need to enable outbound H.323 calls in TERAVoice by selecting the checkbox on the 'Telephony Hardware' tab of the 'Server Configuration' property sheet as described in the topic Telephony Hardware (p. 77).

Set up H.323 Call Handlers
Next step is to create one or more H.323 Call Handlers as described in topic H.323 User Call Handler (p. 99). Configure the assignment settings to determine for which calls this Call Handler should be used for.

Next you need to enter the IP address or the hostname of your H.323 device or the computer where the H.323 application is running on.
In the 'Device (Group)' dropdown list you need to select the device or device group that should be used for outbound calls. If you leave this field empty, TERAVoice will use the devices that are configured for bridged call transfer (see Server Options, p. 91).

Enter a value in the 'Wait for answer' field in order to set the time that TERAVoice will wait for the H.323 device to pick up the call.

If this time is elapsed the unreachable action is carried out. You can for example choose to disconnect the call or redirect to the user's mailbox.

**Permissions**

Permissions are very important for outgoing call functionality in order to prevent people from making calls to numbers they are not allowed to call (e.g. international calls).

You can either choose to allow all numbers except the prefixes in the list or you can choose to allow only numbers that begin with one of the prefixes in the list.

**Setting up Clients**

Next you need to set up your H.323 device or application. In general the only setting you need to make is to set TERAVoice as your gateway for phone calls (not as an H.323 proxy and not as a gatekeeper, these are normally used for calls targeting other H.323 users).

The following examples are for the most common Windows H.323 applications:

**Windows Phone Dialer**
The Windows Phone Dialer (Start ⇒ Run ⇒ Dialer.exe) is using the Windows H.323 TAPI Service Provider. This setting will work for any telephony application that uses TAPI and allows making calls via H.323.
Using TERAVoice - Common Tasks

Just select 'Use H.323 gateway' and enter the IP address of the TERAVoice machine.

**Netmeeting**
There does not exist a Start Menu entry for Netmeeting in WindowsXP but it is still present there and can be started via Start ⇒ Run ⇒ conf.exe.

To configure TERAVoice as a gateway you need select Tools ⇒ Options and then click on the button 'Advanced Calling'.
You need to check the checkbox in the 'Gateway Settings' section and enter the IP address of the machine where TERAVoice is running on.

**USING REMOTE CONTROL FROM A TELEPHONE**

The Remote control mode allows users to control their mailboxes from any telephone. Users can listen to new messages, set mailbox options and record greeting messages. In order to be able to use remote control for a mailbox remote control first has to be configured.

**Configuring Remote Control**

**Server Configuration**

First you need to enable remote control on the Server Options (p. 91) property page.

TERAVoice offers two different methods for authentication in order to prevent unauthorized persons from entering remote control mode.

- By PIN number
  - Allows authentication by entering a number that can be configured for each mailbox

- By CallerID
  - Allows authentication depending on the CallerID (phone number) from which the user calls into his mailbox

You must enable at least one of these authentication methods.

**Mailbox Configuration**

Then you need to enable remote control for the desired mailbox as described in Setting Up Call Handlers Mailbox (p. 96). Enter a PIN number (the PIN number must be at least 4 digits long) or a list of allowed CallerID's or both.
Using Remote Control

To enter remote control mode press '#' ("cross") on the touch tone keypad of your telephone. Depending on the authentication method you now need to enter your PIN or remote control mode starts automatically with if CallerID authentication is enabled and you call from a telephone listed as allowed. You are now listening to a voice prompt telling you the main menu options.

Main Menu

- 1: Listen to messages
- 5: Mailbox options
- 6: Record greetings
- 9: Help information
- *: Quit

Listen to new messages (1)

- 3: Call detail information
- 4: Previous messages
- 5: Repeat message
- 6: Next message
- 7: Delete message
- *,#: Back to main menu

Mailbox Options (5)

- 1: Change E-Mail Notification
- 2: Change SMS/Pager Notification
- *,#: Back to main menu

Record Greetings (6)

- 5: Listen to Greeting Message
- 6: Record Greeting Message
- 9: Enable/Disable Scheduled Greetings
- *,#: Back to main menu

HOW TO CONFIGURE INBAND SIGNALING

Inband signaling is a method an analog PBX can use to inform a voicemail application about the extension number that a call that is signaled at the voicemail port was targeted at. Basic information about inband signaling can be found under Inband Signaling (p. 48).
Activating Inband Signaling

To configure inband signaling you first have to activate inband signaling as a Call Handler Assignment method as described in the topic Call Handler Assignment (p. 63).

Next you need to select the devices you want to use inband signaling with. Open the property sheet of the 'Telephony Hardware' node in the administration console and select the 'Inband Signaling' tab. Check all devices that you want to activate inband signaling for.

For each of those devices you can set several options. When inband signaling is activated for a certain device all 'Wait before taking call' settings that are defined in the Call Handler configurations are ignored and the call is taken after the time that is configured here. You need to specify the maximum amount of time that TERAVoice should wait for the first DTMF tone generated by the PBX and the maximum amount of time to wait between DTMF tones after the first tone has been received.

Setting up You PBX

Of course you need to notify your PBX that you want to use inband signaling on the port connected to the telephony device that you are using with TERAVoice.

This is often done by configuring that port to be of the type 'Voicemail' or 'Phonemail'. Then you need to configure your PBX in a way that this port is either part of a ringing group (so that calls to users are extended after some time to the voicemail port) or to configure the mailbox user’s port in a way that calls are redirected to the voicemail port after a delay.

To verify that inband signaling works you should first connect a telephone to that port and test if you can hear a sequence of DTMF tones right after taking a call.

Pattern Matching

Each PBX is using a different kind of pattern for inband signaling. Information that is usually contained:

- Type of call
  - Can be Direct Internal, Direct External, Redirected Internal, Redirected External
- CallerID
  - Number of the calling party
- Extension number
  - of the mailbox owner

The type of call can be used to enter remote control mode directly in cases where the user calls the mailbox directly (for case Direct Internal). For each type of call the pattern can look different, that is why TERAVoice allows entering 4 different patterns. If your PBX does not support different types of calls you can just use the Redirected External Call field and leave all other fields blank.

How to create a pattern?

A pattern that you enter can contain the following characters:

- 0 1 2 3 4 5 6 7 8 9 0 * #
  - Use one of these characters to specify a digit that must be explicitly present

- X
  - Use this character to ignore one digit
Use this character as a placeholder for the extension number that the call was targeted at. This is the parameter that you need to enter for the assignment rule in the mailbox configuration. The parameter may appear only once within a pattern. The length of this parameter is the length specified above as internal number length.

I do not know how the pattern of my PBX looks like
Documentation of inband signaling patterns is sometimes hard to find but it is always easy to find out the correct settings. Simply enter % as pattern for Redirected External Call and close the configuration dialogs. Restart TERAVoice and be sure that you have already set up a default Call Handler. (To do this create a Call Handler, e.g. a Mailbox, right-click the new entry and select Set as default)

Be sure that you have set up your PBX for inband signaling. Call into your PBX four times in four different ways:
- Call the extension of your voicemail port directly from an internal phone
- Call the extension of your voicemail port directly from an external phone
- Call the extension of a proposed voicemail user from an internal phone
- Call the extension of a proposed voicemail user from an external phone

Now go to the Call Log (p. 108) in the administration console and look for the calls you just made (press F5 for refresh if you cannot see them). Right-click the first entry and select Properties. Look at the list of call log details and search for the inband signaling pattern which should be easy to find. Write down the pattern and remember which type of call this pattern was the result of. Repeat these steps for the three other calls.

Now that you look at these patterns it should not be too hard to find out the logic behind them. If you are unsure you can repeat the procedure and make two more calls to a different internal number.

First replace the internal number with the '%’ character. Then leave all constant characters and replace all characters that might change or are not important to distinguish the type of call with the 'X’ character.

Enter the resulting strings in the 'Inband Signaling’ tab as described above.

Configuring Mailboxes
In order to configure your mailboxes you just need to open the mailbox properties and add an assignment rule on the ‘Basic Options’ page. The rule should contain the internal number of the mailbox user as an entry in the Inband signaled extension field.

Troubleshooting
Always take a look at the call log entries as described above if assignment does not work like expected.

CONFIGURING VOICE PROMPTS
For most voice prompts you can choose to either:
- specify an audio file on the hard disk or
- enter a text that is synthesized via Text-To-Speech
The box for the specification of voice prompts always looks similar to this:

![Voice Prompt]

After clicking on the right-hand button you can either browse for a file on disk or enter a TTS text.

**Note:** You can use any multimedia file that can be played by the Windows Media player (as long as there is no embedded DRM information). For performance reasons (especially when using more than a few lines) it is recommended though, to use only audio files in the format that your telephony device natively supports. Usually this is 8 kHz, 16 Bit, Mono, PCM.

If you enter TTS text this text is always synthesized using the TTS settings specified in Server Options - Advanced (p. 94).
Administration

USING THE ADMINISTRATION CONSOLE

TERAVoice administration is implemented as a SnapIn for the Microsoft Management Console (MMC). You can use the console file that is installed with TERAVoice by starting TERAVoice administration from the start menu.

To do this press Start ⇒ Programs ⇒ TERAVoice Server ⇒ Administration

You can also create your own console that combines several administration tools you want to combine.

Example: Press Start ⇒ Run and enter 'mmc' (without quotes); press enter. After MMC has opened, you click on Console ⇒ Add/Remove SnapIn. Press the Add-Button and scroll down the list until you can select TERAVoice 2004. Press Add. A new dialog pops up asking if you want to administer a local or a remote installation of TERAVoice. Select Local computer or browse to select another computer on the network, then click Finish. Now you can add more SnapIns from the list if you like and close the dialog.

Administration via MMC is done via two different view panes. In the left pane you can see a hierarchy of tree nodes that form a logical structure of configuration options. Many of those nodes offer a set of one or more property pages that can be accessed by right-clicking the nodes or the items in the list views in the right pane if a list view is displayed.
More information about using MMC can be accessed by clicking Help ⇒ Help Topics directly within MMC.

Some changes made to the configuration of TERAVoice require a restart of the TERAVoice server service as described in the following topic When Is A Restart Necessary (p. 76).

**When is a Restart necessary**

Several changes to the configuration of TERAVoice require that the TERAVoice Server service needs to be restarted. Every time you make changes that require a restart, a dialog is displayed asking if you would like to perform the restart.

**Important Note:** When doing a restart of the TERAVoice Server service all current connections are disconnected! Pending notification messages that are not yet sent might be lost!

In order to prevent this from happening you can choose not to restart and open the Server Monitor (p. 107). Make sure that currently no call is connected and that there are no pending messages in the notification queue. Then right-click the monitor node and select Restart TERAVoice Service in order to restart manually.

**Configuration Changes That Require a Restart**

The following configuration options require a restart:

- Changes to the selection of TAPI devices (Server Configuration tab)
- Changes to all Device Settings
- Changes to the Mail Server Settings
- Changes to Inband Signaling Options
- Remote Control settings
- Changes to the Notification Options

**Configuration Changes That Do Not Require a Restart**

The following configuration options will be effective without restart:

- Changes to the Call Transfer settings
- Changes to the Default Settings (remote control voice menu Language, time to take call, max recording time, default greeting messages)
- Text-To-Speech settings (Advanced tab)
- Scripting Options (Advanced tab)
- Remote Control settings
- All changes to any Call Handler configuration

**SYSTEM CONFIGURATION**

**System Configuration**

This topic describes all settings that are available under the Server Configuration node as shown in the following figure:
You can access these configuration options by right-clicking those nodes and selecting Properties.

Note: By selecting the property sheet from the Server Configuration node you can get all the property pages (except Licensing and Version Information) displayed on a single sheet.

Telephony Hardware
To open this configuration dialog please start the TERAVoice administration Console, expand the Server Configuration node and right-click Telephony Hardware to select Properties.

TAPI Devices
This page is divided into two sections. In the first section, TAPI Devices, you can select which TAPI devices you want to use with TERAVoice. Simply check all the devices you want to select.

Note: This list only displays devices that can be used with TERAVoice. If your device is not listed here it does not match the requirements (most likely it does not support audio functions). For details about the requirements for TAPI devices please refer to Telephony Devices (p. 33).

Changes to the selection of devices require a restart (p. 76).

Device Capabilities
If you want to check if your device supports features like Call Transfer simply select the device and click on Device Capabilities. A description of features which may be supported can be found in Device Features (p. 34).

Device Groups
Please refer to the topic Device Groups (p. 37) for more information.

Device Configuration
To configure the driver of your telephony device you can select the device and click on Configure Device. CAPI based devices are shown as ISDN Interface x (Channel y) in this list. More information about configuring CAPI settings can be found under CAPI Configuration (p. 80).

Answer Calls
If your TAPI device is able to report the CalledID (the number that was called, MSN or extension) you can use this option to enter a list of numbers that TERAVoice should operate on. All other calls will be ignored. If you device does not report CalledID information or you do not want to restrict TERAVoice from taking certain calls you need to select Answer All Calls.

Changes to this option require a restart.

Voice over IP Setting
If you want to permit outgoing calls from H.323 users you need to activate this checkbox. This setting does not affect incoming calls (on any of the regular devices) that are transferred to an H.323 target. For more information please refer to How to set up Voice over IP Bridging (H.323) (p. 67).
Device Settings

To open this configuration dialog please start the TERAVoice administration Console, expand the Server Configuration node and right-click Telephony Hardware to select Properties.

Phone Number Modification

Removing Digits
If your telephony devices are connected to a PBX, sometimes the CallerID is transmitted with leading digits that are needed as prefix for outgoing external calls.

Enter the number of digits that should be removed in the Remove leading digits field.

Usually you don’t want to remove digits for internal calls. In order to achieve this you can enter the digit(s) that should be removed in the Only equal to field. You can also enter a pattern with a '?' in this field in order to remove several different prefixes.

Example: To remove '801', '802', '803', '804', '805', etc. please enter 3 and '80?'.

Adding Digits
With ISDN configurations sometimes there is no leading '0' (or other digit depending on the country) transmitted. By activating the 'Add leading digit for external calls' setting, the digit entered into the box will be added for all external calls on this device (a call is considered external when the device driver reports this in the LineCallOrigin parameter).

Changes to these options require a restart.

Audio Delay
Some devices (especially voice modems) require some time to switch to playback mode and therefore voice prompts appear to be playing not right from their beginning but rather some 100s of milliseconds after the actual start of the audio file.

To correct this behavior you can assign a playback delay for those devices. Enter the delay time in the Global audio delay time field and check the Use delayed audio for this device box for each device you want to activate the delay for.

Note: A setting between 400ms and 800ms is recommended for most voice modems.

Changes to this option require a restart.

Call Handler Assignment

To open this configuration dialog please start the TERAVoice administration Console, expand the Server Configuration node and right-click Telephony Hardware to select Properties.

Assignment Settings
This page lets you select which assignment methods you want to use. Check all desired methods. You must select at least one method. For a simple configuration choose the first option By Device, even if you are using only a single device.

Background information about this setting can be found under Basic Concepts - Call Handler Assignment (p. 45).

Changes to this option require a restart.
**Inband Signaling**

To open this configuration dialog please start the TERAVoice administration Console, expand the Server Configuration node and right-click Telephony Hardware to select Properties.

Step by step instructions on how to configure Inband Signaling can be found under How To Configure Inband Signaling (p. 71).

**Configuration Options**

To configure Inband Signaling you need to check each device you want to activate Inband Signaling for in the device list.

You can set up different options for each device although usually the same settings will be used for all devices. To assign the same configuration data for all selected devices simply enter all data and then click on the button below.

*Wait Before Taking Call*

This value determines the amount of time in seconds that TERAVoice waits until it accepts an incoming call on the selected device. This setting overrides the delay settings of all Call Handlers.

Usually you set this value to 0s because with Inband Signaling usually the PBX is responsible for the delay until a call is transferred to the voicemail port.

*Internal Number Length*

In this field you need to enter the digit length of the PBX's internal numbering plan. This is needed for the pattern matching to extract the data from the Inband Signaling pattern correctly.

*Wait For First DTMF Tone*

After a call has been connected TERAVoice waits the amount of time specified here (in ms) after which it stops waiting for the start of a DTMF sequence sent by the PBX.

*Wait Between Tones*

This value specifies the amount of time TERAVoice will wait for another DTMF tone after each tone that has been received. If no more tones are received after this period, the DTMF sequence is considered to be complete.

*DTMF Digit Strings*

There are four text boxes where you can enter a pattern for the following types of call:

- Direct Internal
- Direct External
- Redirected Internal
- Redirected External

A detailed description of how to find out the patterns that you need to use for this purpose can be found under How to Configure Inband Signaling (p. 71).

**Device Groups**

To open this configuration dialog please start the TERAVoice administration Console, expand the Server Configuration node and right-click Telephony Hardware to select Properties.

Click on the Device Groups button to open the dialog for configuring Device Groups:
Use the **New Group** button to create a new group and the **Remove Group** button to delete a group.

All devices that are not assigned to a group are automatically shown under the 'Unassigned' group.

To assign or unassign devices to a group use **Drag and Drop** with your mouse.

**CAPI Configuration**

To display the CAPI configuration dialog you need to open the *Telephony Hardware* (p. 77) property page, select one of the CAPI driven devices (*ISDN Interface x...*) and click on the 'Configure Device' button.

There are two different types of configuration parameters that can be accessed from the main dialog:

- **Main CAPI driver options**
  Select the root node (that corresponds to your computer name) and click on the *Properties* button

- **ISDN Interface Options**
  Select the desired ISDN Interface and click on the *Properties* button

**Important:** The default settings should be fine for most installations. Do not change any settings here unless you want to solve a specific problem.

**Main CAPI driver options**

The ComISDN CAPI driver supports several features that are not used in TERAvoice. You can ignore the settings on the 'Fax' tab, the 'Phone' tab and the 'Compatibility' tab.

Use the 'Driver' tab to display information about the ComISDN CAPI driver version and your ISDN board’s CAPI driver version.

The only relevant settings for use with TERAvoice are those on the 'General' tab:
TAPI Model - Interface

One Line per Interfaces B-Channel: Each ISDN interface’s B-Channel is represented as a single TAPI line. (S0: 2 lines capable of 1 active call each, E1: 30 lines capable of 1 active call each).

One Line per Interface: Each ISDN interface is represented as a single TAPI line. (S0: line capable of 2 active calls, E1 line capable of 30 active calls); this is the most natural representation of ISDN interfaces and is especially suited when using network-based transfer or conference services.

Single line for all interfaces: All ISDN interfaces are represented under a single TAPI line, with one address per ISDN interface. This is another natural representation of ISDN interfaces; its main interest is to enable hardware-based transfer or conference across multiple ISDN interfaces (TAPI enables conferencing and transfer only with calls on the same TAPI line).

Note: What is called a ‘TAPI line’ here corresponds to a ‘TAPI Device’ in TERAVoice terminology.

TAPI Model - MSN

Single Address for all MSN: All configured MSNs will be reported on the same TAPI Device.

One Address per MSN: The number of TAPI Devices resulting from the Interface setting above will be multiplied by the number of configured MSNs.

Note: You should always leave this setting to Single Address for all MSN because TERAVoice has its own features for Call Handler Assignment by MSN!

Network Audio Protocol

This setting decides how audio is encoded in your ISDN network:

- CCITT G.711 µ-law
  This is used in Northern America and in Japan
- CCITT G.711 A-law
  This is commonly used in the rest of the world

**Note:** The default setting is determined by the version of the operating system you are using. If you are using a US operating system version outside of the USA this will result in a wrong value for this setting.

If you are experiencing poor quality or distorted audio, try changing this value.

**Automatic Alerting Usage**
This setting is irrelevant for use with TERAVoice.

**ISDN Interface Options**
Opposed to the Main CAPI Options the ISDN Interface Options are specific to an ISDN interface. If you have more than one board or a board with more than one interface (e.g. 4*BRI/S0) you can set different options for each interface.

The following paragraphs describe only the settings that are relevant for use with TERAVoice.

**General - Interface Type**
There are some H.323 stacks for VoIP that come with a CAPI driver and behave like ISDN boards. In this case you need to select 'IP' instead of 'ISDN'.

**General - B-Channel Connection Mode**
Leave this setting at *In-Band Information/Patterns Available*. This is compatible with most CAPI implementations.

**Features - Tone Monitoring**
This setting decides whether DTMF tones are detected by the ISDN board (hardware) or by the driver (software). You should leave this setting at Hardware and only change it to Software if you are experiencing problems with DTMF detection.

**Features - Hold, Transfer and Conference**
These settings must all be set to the same values, either Hardware or Network. Change this setting to Hardware only if your network or PBX does not support Call Transfer (p. 35). Not all ISDN boards support Hardware transfer (bridging), in this case you can use the Software Bridging Call Transfer mode (p. 55) that TERAVoice offers (which is independent of the modes you can select here).

**Features - Call Transfer**
There are two different modes for Call Transfer (p. 55) in ISDN networks which are referred to as ECT. ECT(E) is the advanced mode that is not supported by many ISDN networks and PBX systems. ECT(I) is the more common mode that is supported by most networks and corresponds to the Network - Implicit setting, which is selected by default.

Check the Enabled on alerting box to allow a call transfer to be completed before the other party has picked up the call. This is necessary to use the TERAVoice transfer mode 'Immediate transfer if not busy' (p. 55).

**Features - DTMF**
These settings determine how DTMF tones are generated and monitored. These settings only apply if you have selected Software mode for DTMF generation and monitoring. These settings will need to be changed only in rare cases.

**Notifications**
To open this configuration dialog please start the TERAVoice administration Console, expand the Server Configuration node and right-click Notifications to select Properties.
Email Notification

The first section allows you to enable or disable the sending of notifications via email. Call Notifications are sent every time a call has occurred while Voice Messages are only sent when a message was recorded.

Click on Edit Template to edit the template that is used to generate the email messages:
You can use all of the field codes from the list below which are then replaced by their corresponding values upon sending of the notification.

In order to get email notifications to work you also have to configure the SMTP-server that will be used; see Server Options (p. 91).

**SMS and Pager Notifications**
Check the box in order to enable SMS and pager notifications. Use Edit Template in order to edit the template for these notifications as described above.

**Sender Number**
Use this box to specify a sender number. This number should correspond to a TERAVoice Remote Control Call Handler (p. 103) to allow users to call back the sender to gain direct access to the remote control mode.

**SMS/Pager Configuration**
Click on this button to configure the SMS/Pager component. The configuration settings are explained in the next topic, SMS/Pager Configuration (p. 86).

**PBX Notification**
Check the box to activate PBX notification via the MWI lights on PBX telephones. For basic information about MWI notification please refer to the topic Message Waiting Indication (MWI) (p. 54).

**Numbers For Activation**
Use these boxes to enter the patterns that TERAVoice should dial in order to activate or deactivate the MWI light on a specific phone. Use ‘%’ as a placeholder where the number is inserted that is specified in the corresponding mailbox’s station for MWI notification (see Mailbox Configuration (p. 70)).

**Retries And Interval**
Enter the amount of retries here in order to perform PBX notification. The interval setting specifies the amount of times between retries.

**TAPI Device**
Please select the device that TERAVoice should use to dial the code for activation or deactivation here.
**Important note:** If you are running a voicemail application with a high volume of calls you should use a device here that is not used for incoming calls (or at least is very unlikely to receive incoming calls). If TERAVoice tries to dial the PBX notification sequence and the device is occupied with an incoming call the attempt to dial out may fail (after the specified amount of retries) and the PBX notification will not get executed. This behavior also depends very much on the number of lines that a TAPI device supports: If you have one device that offers 30 lines you won’t run into problems but if you have a board that uses one TAPI device for each line (e.g. 30 TAPI devices), this problem may occur.

**SMS/Pager Configuration**

To open this configuration dialog please start the TERAVoice administration Console, expand the Server Configuration node and right-click **Notifications** to select **Properties**. Then click on the button **SMS/Pager configuration**.

**Services**

The first thing you need to do here is to set up a service you want to use for sending your SMS or pager messages.

Click on the services tab to display the list of configured services:

![SMS Configuration dialog](image)

To add a service click Add and select a predefined service or add a new service. If you add a new service you need to know the configuration information necessary to sent messages with your GSM or paging provider:
We may provide updates of the list of predefined services at irregular intervals. Please visit the following web-page for up-to-date information about SMS and paging services:


If there is a new file with predefined services available you can use the *Load from file* button in order to use this new information.

**Service Configuration**

If you have selected a predefined service usually no further configuration is necessary. To specify settings for a new service or to change the settings of a predefined service click on the *Properties* button below the service list to display the Service Settings dialog:
**Country Code**
Enter the country code that this service's numbering plan belongs to.

**Protocol Type**
Enter the type of protocol that is used by your SMS or paging provider. You can choose between TAP and UCP.

**Dial-Up Number**
Enter the number that you need to dial to connect to your provider's access point. You should use the canonical format, so that Windows dialing rules can be applied properly: +country (area) number

**Password**
Enter the password for your paging provider if you are using the TAP protocol.

*Note:* The password is PG1 in most cases

**Max Message Length**
Enter the maximum length that is allowed for messages. This setting depends on your provider. For SMS networks this is usually 160 characters. With paging providers this value differs.

**Max Messages Per Session**
Enter the maximum amount of messages that you are allowed to send without disconnecting first. If you are not sure set this value to 1 and try increasing it later.

**UCP Operation**
The UCP protocol offers different methods for submitting messages. If you are unsure you should try 01 first, then 51 and then 30.

**Message Type**
Some paging services only allow numeric messages to be sent. In all other cases you can select Alphanumeric.

*Note:* If you are using a numeric only paging service you need to make sure that the notification messages sent only contain numbers by editing the notification template as described in topic Notifications (p. 83).

**Timeout**
Specify the amount of time in ms to wait for a response from the server.

*Note:* Some paging services take some time to respond because the connection to their service is used for charging for message submissions. If you think sending messages fails because of a timeout issue you can try to increase this value. The default value is 2000ms.

**Baud Rate**
Specify the baud rate your modem should use to connect. If you do not get a connect with the Auto setting you can try 1200 or 9600 which should work in most cases.

**Data- Parity and Stop-Bits**
The most common setting here is 8N1 while some services require 7E1.

**Modem Device**
If you are using more than one service you can select a modem to use with this service. Otherwise you can select Default and select your modem on the Common tab of the configuration dialog.

**Common Settings**
On the common settings dialog the following options can be selected:
Default Transport
Select the service here that should be used if none of the rules specified on the Routing tab applies. Routing rules are only required if you are using more than one transport service.

Default Modem
Specify the modem here that should be used by default with all configured transport services. You can override this setting by specifying a modem in the configuration dialog of each transport service as described above.

Note: We currently only know of two GSM providers in Germany that offer a dial-up access point that can be reached via ISDN. All other services require analog modems to be used. There are ISDN boards that come with an analog modem emulation. During our tests we often encountered problems with these emulated modems because they have problems connecting on low baud rates. Therefore we recommend always using a normal analog modem for sending SMS and pager messages unless you are using one of the two German ISDN access points.

Replacement Operations
Some SMS/paging gateways have problems with the '+' character. Use the first option to replace this character with '00'.

The second option is used to add the country code to numbers that do not have country code (i.e. numbers that start with '0' but not with '00'). To add the German country code you can check this box and enter '0049' in the text box.

Retries For Sending
Enter the amount of retries to send a message. The default value is 3.

Retry Delay
Enter the amount of time in minutes TERAVoice should wait between retries.

Routing
If you are using more than one transport service you need to determine which service should be used. This is done depending on the number of the recipient of the SMS or pager message:
Press Add to add a prefix number. Enter the number and click OK. Then click on Properties and specify the service that should be used to send messages with this prefix.

For all numbers that have a prefix which is not on this list, the default transport service (as specified on the Common tab) is used.

**Diagnostics**
Finally you can test your configuration by using the test-tool which is available from the Diagnostics tab.

**Test-Tool**
Enter the sender and recipient numbers and the text of the message to send. Press the Send button and watch for the information that appears in the lower window. In most cases this information will help you to find the problem if sending a message fails.

**Dialing Location**
This button is a short-cut to opening the Windows dialing rules configuration dialog that is also available from the Windows control panel. Use this dialog to control how numbers (of the service providers) should be dialed (like adding a digit to get an outside line if behind a PBX or to strip or add area codes, etc.).

In order to get these dialing rules to work the dial-in numbers must be entered in canonical format (see Service Configuration above).

**Licensing and Activation**
To open this configuration dialog please start the TERAVoice administration Console, expand the Server Configuration node and right-click Licensing to select Properties.

**License types**
This dialog lets you view, add, remove and activate your licenses. There are three different types of licenses available for TERAVoice:

- **Base licenses**
  Each installation needs a basic product license which includes a certain number of lines and Call Handlers. Only one base license may be entered for each installation.
- **Lines expansion licenses**
  With lines expansion licenses you can add more lines to the number of lines that is available with the effective basic license. Several lines expansion licenses can be entered.

- **Call Handler expansion licenses**
  With Call Handler expansion licenses you can add more Call Handlers to the number of Call Handlers that is available with the effective basic license. Several Call Handler expansion licenses can be entered.

**Activation**

Every license needs to be activated. A license that is not activated will not be effective. Please click on the Add button first and enter your serial number.

You can find your serial number on the license certificate. The serial numbers for expansion licenses can be found on the license certificate.

If you have purchased your product online you will find the serial number on the receipt of purchase you have received via email.

There are two types of activation methods:

**Online Activation**
This is the easiest method for activating your product. Click on the Activate License button and select Online Activation. Follow the instructions on the screen to complete the activation process.

**Important:** You need an active internet connection to use online activation!

You have the option to register your product during activation by entering your personal details. If you want to submit a support request, you need to be a registered user.

**Note:** It is not required to submit your personal information in order to activate your product!

**Email Activation**
If you do not have an internet connection or you are unsure about the data that is transmitted during activation you can use email activation. Click on the Activate License button and select "Email Activation", then choose "Create Request file". The software will then let you choose a location to save a file called tvLicReq.txt. Send this file via email to teravoice@registration.terasens.de. The TERASENS license server should usually respond within a few minutes. If activation was successful the response message will contain another attachment called tvLicResp.txt.

Now open TERAVoice Administration again and proceed to the license dialog. This time select "Process Response File" and specify path and location to the tvLicResp.txt file to complete activation.

**Troubleshooting**
If you have any problems with activation please refer to the topic If Activation Fails (p. 148) in chapter Troubleshooting And Support.

**Server Options**
To open this configuration dialog please start the TERAVoice administration Console, expand the Server Configuration node and right-click Server Options to select Properties.

**Remote Control**
This page lets you set several options for remote control:

*Allow Remote Control*
Activates or deactivates Remote Control system-wide.

**Authentication Modes**
TERAVoice offers two different methods for authenticating users trying gain access to the Remote Control mode.

- **By Remote Control Code**
  The user can access Remote Control by entering a PIN number

- **By CallerID**
  The user can access Remote Control if he is calling from a telephone that has a number (CallerID) that is in the list of allowed CallerID's

**Note:** At least one of these authentication methods must be activated, otherwise Remote Control will not work.

**Autostart Remote Control Mode**
If a user is authenticated by his CallerID you can select here that Remote Control mode is automatically entered without the need for pressing '#'.

For more information about configuring and using Remote Control please read Using Remote Control From A Telephone (p. 70).

**Event Logging**
TERAVoice offers 4 different methods for logging information about errors. There are 4 different log levels available:

- **Errors**
  Logs error messages only

- **Errors and warnings**
  Logs error and warning messages

- **All information**
  Logs error, warning and informational messages

- **Debug**
  Logs additional information for tracking errors

Not all logging messages allow logging with any log level. For example debug information is only available when logging to file.

**Note:** Logging debug information can affect your system performance. Please use this option only for tracking down errors.

**Internal Log**
TERAVoice has its own internal log file to record information about system events. This information can be easily viewed within the TERA Voice administration console as described under Monitoring System Activity (p. 107).

This log is also available for remote administration.

**Log To File**
This option is useful for error tracking. When submitting support requests you may be asked to create a file log with debug level information.

**Log To System Event Log**
The Windows Event Log provides a central instance where information from all system services and applications is stored. Therefore it is a good practice to activate logging to the Windows Event Log.

Send Via Email
In order to get informed about errors in TERAVoice most quickly you can select to send error messages to your email address.

**Note:** In order to receive error messages via email you need to correctly configure the SMTP server settings as described under Server Options (p. 91)

**Recommended Settings**
The recommended settings for logging are:

- Internal Log: All Information
- Log To File: None
- Log To System Event Log: Errors and warnings

**Call Transfer**
This page lets you set the options for Call Transfer. Basic information about transferring calls with TERAVoice can be found under Call Transfer Types (p. 55).

**Transfer Mode**
Select the Transfer Mode you want to use with TERAVoice. A description of all transfer modes can be found under Call Transfer Types.

**Wait Until Answer**
Select the amount of time TERAVoice should wait for the C-party to take the call. If the call is not picked up within this time the transfer will be considered to have failed.

**Always Simulate Hold**
This setting only applies to bridged transfers. You can specify an audio file below that is played back to the A-party instead of putting the A-party on hold.

If putting the A-party on hold fails (e.g. because the device or the network does not support hold) the Simulated Hold Audio file is played back anyway, but if you select this option TERAVoice never tries to put the call on hold.

**Simulated Hold Audio**
Please enter an audio file that should be played back as hold music for simulated hold during bridged Call Transfer.

**Use These Devices For Bridged Transfer**
When doing normal Call Transfer the transfer is executed as a network or PBX function. For simulated transfer a completely independent connection to the C-party is created. Any device can be used for this second connection (unless it does not support full-duplex mode, such as most modems).

Select all devices here that should be used for creating the C-party connection.

**Options**
The Options page lets you configure several basic and default settings.

**Mail Server**
Enter the data of the SMTP server here. You can enter an IP address or the DNS name of the mail server you want to use.

Enter the display name and the email address that should be used as the sender for all email messages that are sent by TERAVoice.
Click on the Advanced button in order to specify additional settings like logon information or server port.

**Default Settings**

**Voice Menu Language:** Enter the default language that is to be used for the voice prompts in Remote Control (p. 42) mode.

**Wait before taking call:** This is the default time until TERAVoice connects an incoming call.

**Max recording time:** This is the default maximum duration for recorded messages.

*Note:* Those settings can be overridden by settings in every Call Handler’s configuration options.

**Default Messages**

**Goodbye Message:** This is the default message that is played if the maximum recording time is reached.

**Start-Recording Sound:** This is the sound that is played back as a signal that a user can start leaving a message.

*Note:* Those settings can be overridden by settings in every Call Handler’s configuration options.

**Advanced**

**Scripting Options**

**Script Timeout:** You can specify a timeout value here after which an IVR Script is considered to be “hanging” and terminated.

**Script Debugging:** Use this setting to enable debugging of IVR Scripts.

*Important:* Use Script Debugging only for test purposes. If you enable debugging all scripts will be handled by the same thread and other scripts will hang if you are currently debugging a script! When debugging scripts always set the Script Timeout to 0s (=disable!)

**Text To Speech**

**Call-Info Mode:** When using normal (non-TTS) mode TERAVoice can only announce the CallerID but not the date and time of a call. Use this setting to enable TTS for Call-Info announcement including date and time.

*Note:* You can press 3 when listening to a message in remote control mode to play the Call-Info announcement.

**TTS Voice:** Select the voice for Text To Speech that you want to use.

More information about Text To Speech can be found under Text-To-Speech (p. 54).

**Version Information**

To open this configuration dialog please start the TERAVoice administration Console, expand the Server Configuration node and right-click Version Information to select Properties.

This dialog allows you to view the version details of all relevant files of your TERAVoice installation.
If you are going to submit a support request you may be asked to report the information found here.

**Tip:** Click on the Check for update button in order to check if there is a newer version available.

### SETTING UP CALL HANDLERS

This topic describes the settings for all Call Handlers that are available in TERAVoice. To manage Call Handlers you can expand the Call Handlers node on the tree-view pane. Click on the Call Handlers node to display a list of all Call Handlers. If you want to display a subset of configured Call Handlers you can click on one of the nodes below to display Call Handlers of the specified type.

#### Basic Options

To display this page open the property sheet of any Call Handler by right-clicking and selecting Properties.

This page is common to all Call Handlers and contains basic options like name, ID, description and assignment options.

#### General

**Call Handler ID**

The ID uniquely identifies a Call Handler. While you can change name and description at any time the ID cannot be changed in this dialog.

If you want to change the ID of a Call Handler for some reason you need to right-click the Call Handler in the list and select **Change ID**.

**Important Note:** If you change the ID of a Call Handler all Call Log information will be lost. All voice menus or other Call Handlers that were set to hand over call handling to this Call Handler will fail after the ID is changed.

**Name**

In this input field you can enter a name for the Call Handler. You can change this name at any time you want without impacting any kind of functionality.

**Description**

You can enter a description for the Call Handler in this input field to store additional information associated with this Call Handler.

**Type**

This input field displays the type of the current Call Handler. This field cannot be changed.

**Call Handler Assignment**

In order to inform TERAVoice for which calls the current Call Handler should be used you need to enter one or more assignment entries in the list.

You can leave the list blank if this is the default Call Handler or you do not currently want to use this Call Handler for incoming calls.

**Note:** Another reason to leave this list blank could be that you there is forward processing to this Call Handler from another Call Handler but this Call Handler is never used to initially process incoming calls.
Adding Assignment Entries
To add an entry to the list, please press the Add button. A new dialog pops up which lets you enter parameters for all types of assignment methods you have activated on the Call Handler Assignment (p. 78) configuration page.

You can enter an unlimited number of entries in this list. You can use Device Groups (p. 37) instead of individual devices (as the device parameter) in order to reduce the number of entries necessary if you are using a larger number of devices.

**Note:** More detailed assignment entries always take precedence opposed to less detailed entries.

**Example:** If you have activated 3 assignment methods and you have configured an assignment entry with the exact and matching information and you have another Call Handler with an entry where 2 parameters match and the third is set to Any, then the entry with 3 parameters will take precedence.

More information on Call Handler Assignment can be found in the chapter Basic Concepts (p. 45).

Default Call Handler
There is always exactly one Call Handler set as the default Call Handler. The default Call Handler will take all calls that do not match the assignment entries of any configured Call Handler.

To set the current Call Handler as default, simply click on the Set As Default button.

Call Options
Time To Take Call
You can enter an amount of seconds here that TERAVoice waits until the Call Handler answers the call. If you select the Use Default checkbox, TERAVoice uses the value that is configured in the Server Options (p. 91) dialog.

**Note:** This option is not available for the Music On Hold Call Handler. This Call Handler always uses a delay of 0s.

Mailbox Call Handler
This topic describes the configuration pages for Call Handlers of the Mailbox type. To open the property sheet containing these pages right-click on a Mailbox Call Handler and select Properties.

Basic information about this Call Handler type can be found under Call Handlers - Mailbox.

The options on the General page are described in the previous topic, General Options (p. 66).

The following property pages are available for this Call Handler type:

- General (p. 95)
- Message Schedule
- Notifications
- Options
- Security

Message Schedule
Each mailbox can have one ore more greeting messages that can be configured. By default only one message is active and message schedules are disabled.

Voice Prompt
Select the greeting message you want to have played back for the selected schedule or played back always if there is only one message schedule defined. Help on setting voice prompts can be found under Configuring Voice Prompts (p. 73).

**Note:** If you leave the field for the message empty, the default message as defined in Server Options (p. 91) is used.

**Recording**
Select the checkbox Allow Recording if you want to allow callers to leave a message. If the box is unchecked, the call will be disconnected after the message has been played.

**Configuring Schedules**
You can add any number of schedules to the list by clicking the Add button. Select all restrictions in date, time and weekday you want to apply for the selected schedule.

**Important:** When executing a Call Handler with scheduled greeting messages TERAVoice will always start with the schedule that has the highest priority (i.e. highest number in the right column in the schedule list). If the schedule criteria match the current date and time this schedule is used. If not TERAVoice will continue with the schedule that has the next lower priority value until it finds a match or it comes to the default schedule which is then used.

Use the Move Up and Move Down buttons to change the priority position of the selected schedule.

**Notifications**
Use this page to configure the notification options that should be used for this mailbox. All notification types will only work if they have been activated and configured on the Notifications (p. 97) page in the system configuration.

**E-Mail Notification**
Select the checkbox and enter a valid email address in order to activate email notifications for:

- Voice Messages
  This notification is only sent if a message was recorded
- Notification Messages
  This notification is sent every time a call has occurred

**Note:** If both options have been selected and point to the same email address only one notification is sent if a message was recorded.

**SMS/Pager Notification**
Enter the number of the pager or the mobile phone that the notification message should be sent to.

If you select the first checkbox you can choose to only send notifications if a message was recorded.

The default sender number setting specified in the SMS/Pager Configuration (p. 86) can be overridden here to allow the user to call back into his mailbox. If no Remote Control Call Handler (p. 103) is configured on a global extension or phone number you can use this setting to enter the number to call directly into the user’s mailbox.

**MWI Notification**
Check the box and enter the extension number of the phone whose MWI light should be used for notifications about new messages in this mailbox.
Options
This page lets you configure several mailbox options.

Max Recording Time
Enter the maximum amount of time that TERAVoice allows for recorded messages (by the caller). Select the Use Default option in order to use the value that is configured in Server Options (p. 91).

Goodbye Message
Select a message that is to be played to the caller after the amount of time specified in the previous setting is reached. Select the Use Default option in order to use the value that is configured in Server Options.

Tip: It would certainly be helpful to the caller if the message were to indicate something like: "The maximum amount of recording time is reached. Thank you very much for your call."

Menu Language
Select the language of the voice prompts that TERAVoice uses for Remote Control (p. 42) mode. Select the Use Default option in order to use the value that is configured in Server Options (p. 91).

Remote Control
Select the checkbox in order to allow this mailbox be accessed via Remote Control. Depending on the authentication modes that you have allowed on the Remote Control Configuration Page (p. 91) you can enter a PIN number and/or a list of CallerID’s that are allowed to access the mailbox in Remote Control mode.

Security
This page is the standard Windows security page. Here you can set access permissions for the folder all messages for this mailbox are stored in. You could also use the Windows Explorer to set these security settings. The page is only provided for reasons of convenience.

More information on setting the security to allow users accessing their messages over the local network and the folder structure that is used to store messages can be found under Security Considerations (p. 59).

Voice Menu Call Handler
This topic describes the configuration pages for Call Handlers of the Voice Menu type. To open the property sheet containing these pages right-click on a Voice Menu Call Handler and select Properties.

Basic information about this Call Handler type can be found under Call Handlers - Voice Menu (p. 39).

The options on the General page are described in the previous topic, General Options (p. 66).

The following property pages are available for this Call Handler type:

- General (p. 95)
- Menu Actions
**Menu Actions**

**Voice Message**
Select the message (or other kind of audio file) you want to have played back right after the call has been connected. Help on setting voice prompts can be found under Configuring Voice Prompts (p. 73).

**Actions List**
Click on the Add button in order to add an entry to the list. There are two different types of events that can be added:

- **Digit Events**
  A digit event occurs when the user presses a certain key on his touch-tone keypad

- **Timeout Events**
  A timeout event occurs after a specified amount of time

**Note:** There is also a special digit event "All other digits" that occurs if a key not defined in the list is pressed. If you do not add this type of event nothing happens if the user presses a key that is not in the list.

Select the desired type of event and click OK to add the entry to the list. The property dialog is then shown to let you choose the type of action that is to be carried out when this event occurs. The following action types are available:

- **Go to Call Handler**
  Hands processing over to another Call Handler

- **Repeat message**
  Play back the audio message specified above once again

- **Hangup Call**
  Disconnects the call

- **Transfer Call**
  Starts a Call Transfer to another telephone number or to a H.323 (VoIP) user. Please refer to the topic Call Transfer Types (p. 55) for further information. If you want to transfer to a H.323 User (p. 40) you need to select Bridged Transfer.

**H.323 User Call Handler**

This topic describes the configuration pages for Call Handlers of the H.323 User type. To open the property sheet containing these pages right-click on a H.323 User Call Handler and select Properties. The options on the General page are described in the previous topic, General Options (p. 66).
**Common**

*IP/Hostname*
Enter the IP address or the host name of the computer, IP phone or other H.323 device (=H.323 terminal) that this Call Handler is assigned to. Incoming calls that match the assignment (p. 45) parameters of this Call Handler are automatically signaled at the H.323 terminal and transferred (bridged) as soon as the call is accepted.

If the call is not taken on the H.323 terminal within the duration specified in the 'Wait for answer' field, the **Unreachable Action** is executed.

*Device/Group*
In the *Device/Group* drop-down field you can choose which devices can be used for outgoing calls.

*Outgoing MSN*
If you are using a CAPI based ISDN Device you can select the outgoing MSN that TERAVoice sends as the Caller ID to the network.

*Gateway/Gatekeeper*
If the H.323 Call Handler is used to serve users behind a Gateway (like Cisco CallManager) or a Gatekeeper, you must enter the IP address of the gateway for all users in the IP/hostname field. The number that the user has been assigned within the numbering plan of the Gatekeeper/Gateway needs to be entered into the *Target Number* field.

To be able to correctly connect inbound calls to the Gateway/Gatekeeper system the IP address of the Gateway/Gatekeeper must be entered in the Windows H.323 configuration. Please press the button to enter this information.
**Unreachable Action**
When the H.323 terminal does not take the call within the time specified above then the Unreachable Action is carried out. You can choose from:

- Switch to Call Handler
- Transfer Call
- Hang up Call

**Dial Out Permission**
In order to prevent users from calling certain numbers or to restrict the numbers they are allowed to dial to a certain set you can use the Dial Out Permissions.

Choose between the 'Allow all except...' and 'Allow only...' and add the numbers to this list.

To completely disallow outgoing calls select the second option and leave the list empty. To allow all kinds of outgoing calls you can choose the first option and leave the list empty.

A simple example for how to allow local calls and long distance calls but no international calls would be to choose the first option and add '000' as an entry to the list.

**Note:** The code to dial international numbers may be different in your country.

You may also want to disable certain call by call providers or certain service numbers with high rates: Simply add the unwanted prefixes to the list.

**Music on Hold (MOH) Call Handler**
This topic describes the configuration pages for Call Handlers of the Music On Hold type. To open the property sheet containing these pages right-click on a Music On Hold Call Handler and select Properties.

Basic information about this Call Handler type can be found under Call Handlers - Music On Hold (p. 101).

The options on the General page are described in the previous topic General Options (p. 66).

The following property pages are available for this Call Handler type:

- General (p. 95)
- Music On Hold

**Music On Hold**

*Connection Mode*
Select the desired connection mode for this Music On Hold Call Handler. If you select the second mode you need to supply a phone number that is to be dialed and the TAPI Device that TERAVoice should use to connect.

The connection modes are described under Call Handlers - Music On Hold.

*Voice Message*
Select the audio file that is to be played as on hold music. The audio file is looped infinitely until TERAVoice is shut down or the PBX closes the line.

**IVR Module Call Handler**
This topic describes the configuration pages for Call Handlers of the IVR-Module type. To open the property sheet containing these pages right-click on an IVR-Module Call Handler and select Properties.
Basic information about this Call Handler type can be found under Call Handlers - IVR Module.

The options on the General page are described in the previous topic, General Options (p. 66). The following property pages are available for this Call Handler type:

- **General** (p. 95)
- **Script**
  - If you have selected an IVR Module of the Script type
- **COM Component**
  - If you have selected an IVR Module of the COM Component type

**Script**

**Script Language**
The IVR Module of the Script type lets you control TERAVoice either via

- **VBScript**
  - Microsoft Visual Basic Script or
- **JScript**
  - Microsoft’s implementation of JavaScript

You can use different scripting languages like Perl or Python but these languages are not officially supported. In order to use different scripting languages you need to have this language installed and enter its name in the Script Language field.

**Open Script Editor**
Click this button to open the integrated Script Editor. With the Script Editor you can load and save your script files from and to files.

**Note:** The active Script is always stored in the TERAVoice configuration database.

You can find documentation on the Script Editor in the Script Editor (p. 128) topic.

**Check Declaration of Functions**
With Scripting no design-time error checking or compilation is done. To check for the most common errors because of invalid or missing function declarations you can click on this button to check the current script.

**COM Component**
This page appears for IVR modules of the COM Component type. Enter the ProgID of your COM (ActiveX) component here and press the *Check Interfaces* button in order to check if

- the component can be instantiated properly and
- if it supports the required interface.

Further information about using the IVR Module Call Handlers can be found in the chapter Developing IVR Applications (p. 115).

**Scheduled Execution**
Usually IVR Scripts or COM components are executed to handle incoming calls. TERAVoice provides a method for scheduled execution of IVR Call Handlers, too. This can be used for performing certain maintenance tasks that need to be executed regularly and it can be used for creating outbound calls.
Instantiation Settings
Select the checkbox to enable scheduled execution of your IVR Call Handler. With the settings below you can control how instances of your IVR Call Handler are created.

With the value in the 'Execute every' setting you can set the interval how often the Call Handler is instantiated. You can also set the maximum amount of simultaneous active instances of the Call Handler. For applications that need to perform a large number of outgoing calls this is very useful. The limit applies only to scheduled executions of the Call Handler; there is no limit for incoming connections.

**Note:** Every allowed instance is counted as a Call Handler and will affect the number of licensed Call Handlers. If you have a license with 5 Call Handlers you can enter a number of 5 as a maximum, given that there are no other Call Handlers configured!

With the instantiation mode you can control how instances are created: Either one instance maximum is added every interval or TERAVoice can add as many instances of the Call Handler as are permitted on every interval.

You can limit execution to certain days of week, periods of time or certain dates with the control on the page.

IVR Handler Log Event
You can additionally have your IVR Module executed every time a call log entry is generated. This is useful for certain applications where accounting information needs to be processed or you want to create an outbound call in order to inform a user about a certain call.

Remote Control Call Handler
This topic describes the configuration pages for Call Handlers of the Remote Control type. To open the property sheet containing these pages right-click on a Remote Control Call Handler and select Properties.

Basic information about this Call Handler type can be found under Call Handlers - Remote Control (p. 103).

The options on the General page are described in the previous topic General Options (p. 66).

The following property pages are available for this Call Handler type:

- General (p. 95)
- Remote Control

**Remote Control**

**Options**
Select the desired options for this Remote Control Call Handler.

Check the Auto-Login setting if you wish to automatically log in users that are calling from CallerID’s that are allowed for Remote Control for exactly one mailbox (not 0 and not more than one).

Check the Ask for MailboxID if you want to allow users to log in with their mailbox number and their PIN code.

The "Don't require PIN on matching CallerID" setting allows users (calling with a CallerID that is allowed for more than a single mailbox) to enter only their mailbox number but not the PIN code.

**Menu Language**
Select the language of the voice prompts for this Remote Control Call Handler. You can use this setting to implement several Remote Control Call Handlers with different languages.

Usually you should check Use Default Language.
**Time Schedule Call Handler**

This topic describes the configuration pages for Call Handlers of the *Time Schedule* type. To open the property sheet containing these pages right-click on a *Time Schedule* Call Handler and select *Properties*. Basic information about this Call Handler type can be found under *Call Handlers - Time Schedule* (p. 42). The options on the General page are described in the previous topic General Options (p. 66).

The following property pages are available for this Call Handler type:

- **General** (p. 95)
- **Schedule**

**Schedule**

**Schedule Settings**

You can add as many schedules as needed to the list by clicking the *Add* button. Select all restrictions in date, time and weekday that you want for the selected schedule to apply.

---

**Important:** When executing a *Time Schedule* Call Handler TERAVoice always starts with the schedule that has the highest priority (i.e. highest number in the right column in the schedule list). If the schedule criteria match the current date and time this schedule is used. If not TERAVoice continues with the schedule that has the next lower priority value until it finds a match. If no match is found then the call is not answered or disconnected if the call was already in connected state.

Use the *Move Up* and *Move Down* buttons to change the priority position of the selected schedule.

**Action Settings**

Select the action that is to be carried out for the selected schedule. The following actions are available:

- **Switch to Call Handler**
  Switches to the selected Call Handler to continue processing

- **Transfer Call to**
  Initiates call transfer to the selected target

- **Hangup Call**
  Disconnects the call

---

**Note:** You can not select VoIP targets here for Call Transfer. Instead you can create a H.323 Call Handler and select to switch to this Call Handler.

**Call Answering**

---

**Please remember:** The *Time Schedule* Call Handler is the only Call Handler that can act on calls that are either connected or not yet connected. For non-connected calls the further behavior (e.g. Time to take call) is determined by the Call Handler that the *Time Schedule* Call Handler switches to. The *Time Schedule* Call Handler never answers calls that are in ringing state, except for when the selected action is Transfer Call to.
Procedure where A calls B. B creates a consultation call to C while holding A. B then connects/transfer A to C.

**Waiting Queue Call Handler**

This topic describes the configuration pages for Call Handlers of the Waiting Queue type. To open the property sheet containing these pages right-click on a Waiting Queue Call Handler and select Properties.

Basic information about this Call Handler type can be found under Call Handlers - Waiting Queue (p. 105).

The options on the General page are described in the previous topic, General Options (p. 66).

The following property pages are available for this Call Handler type:

- General (p. 95)
- Waiting Queue
- Agents
- Queue Viewer

**Waiting Queue**

*Common Options*

If you want a welcome message to be played you can select an audio file or a TTS prompt here in the first text box.

The available Cycle Modes are described under Call Handlers - Waiting Queue (p. 31).

Enter the time TERAVoice should wait until an agent picks up the phone in the Wait for agent text box.

*Recording Conversations*

Check this box if you want to record conversations automatically. The following requirements need to be fulfilled in order for this feature to work:

- Your PBX or public network must support 3 Party Conference
- Your telephony board must support 3 Party Conference
- Your telephony board's TAPI driver must support 3 Party Conference
  (if you are using a CAPI based ISDN board the board's CAPI driver must support this feature; most CAPI drivers do so)

If one of these requirements is fulfilled then the Call Transfer will fail!

**Note:** This feature allocates one line of your telephony board (and of course of your PBX/network) for each conversation until the conversation is finished.

*Queue Full Condition*

You can specify a maximum number of slots that are available for this Waiting Queue. This is helpful to limit the available resources (lines) and to avoid excessive waiting times before connection occurs.

You can optionally specify a message that is played back when the queue is full and you need to specify an action that should be carried out in this case. The following types of actions are available:

- Switch to Call Handler
  Switches to the selected Call Handler to continue processing
- Transfer Call to 
  Initiates call transfer to the selected target
- Hangup Call 
  Disconnects the call

Playback During Wait
During waiting time the call is put on hold in order to have the caller listen to the Music On Hold that the PBX (or the public network) generates (you can also specify to simulate Hold instead; see Server Options - Call Transfer (p. 93)).

You can specify a message here which is played in selectable intervals: The call is retrieved from hold, the message is played and the call is put on hold again.

Announce Position On Change
If you select this option a message is played back to the caller each time his position in the queue changes. The message is: "You are now position x on the waiting queue."

Agents
The list should contain all agents that could be active for a certain waiting queue. An agent can easily set to 'active' or 'inactive' by clicking on the 'Toggle Active' button.

For some Cycle Modes the order of the agents is important. You can use the arrow buttons on the bottom right to move certain agents up or down in the list.

When you add a new agent or edit an existing one you get the same dialog as for normal call transfer. Please refer to Call Transfer Types (p. 55) for a description of the available transfer modes.

Note: The recommended transfer mode for waiting queues is "Wait until connect, then transfer" or "3 Party conference" if you want to record conversations.

Queue Viewer
With the list on the 'Waiting Queue Viewer' tab you can view the current state of the waiting queue with all calls that are currently waiting and their positions on the queue. The list is refreshed every second.
Monitoring System Activity

SERVER MONITOR

This topic describes the Server Monitor. To display the Server Monitor expand the Monitor node on the tree-view pane. Click on the Server Monitor node to display the Server Monitor.

The Server Monitor displays all current activity. It is divided in two parts. The upper section displays all calls that are currently connected. The lower pane displays several types of information, depending on the tab that is currently selected.

Start and Stop Server

If you can see a message indicating that TERAVoice is not running you can start TERAVoice by right-clicking on the Server Monitor node and selecting Start.

You can also stop or restart TERAVoice in this manner.

Active Calls List

The upper pane displays a list of calls that are currently connected. The maximum number of entries in this list corresponds to the number of lines that the active license allows.

Please note: There is no direct correspondence between physical lines or logical lines or TAPI Devices to the entries in this list.
This matches the TERAVoice licensing scheme. The maximum number of lines that a license allows always refers to the number of simultaneously connected lines. You can use a modem, an ISDN board and an analog telephony board even if you only have a license for one line! However, then only a single connection at a time can be made, meaning that other calls during this connection are not taken.

You can see the status for each entry in the list, including the time that a call has been connected, the Call Handler that is currently active and the current state for mailboxes, Remote Control Call Handlers etc.

Detail Information

Call Details
In this list you can see the details log of the call that is selected in the upper pane. You can instantly see all actions that are taken, messages that are played or recorded and digits that the caller presses.

This information is the same as that which can be found in the details information in the Call Log (p. 108).

Notify Queue
Notifications are not executed immediately but are put in a queue. This is done to serialize notification handling and to free up the Call Handler to be ready for a subsequent call. The list shows all pending notifications in the queue.
**Tip:** It is a good idea to have a look at the notification queue before stopping TERAVoice Server. When stopping TERAVoice all notifications are erased!

**SMS Log**
This list shows the current queue of the SMS/pager sending component. The SMS sending component has its own queue for sending messages. As soon as an SMS notification is processed by the Notify Queue it is submitted to the SMS sending component.

**Note:** The SMS sending component is running out-of-process (tsSMSQueue.exe). You can shut down TERAVoice without affecting messages that have already been submitted to the SMS queue.

**Call Log**
This list contains the same information as the main Call Log. Only calls that have occurred since the last startup of TERAVoice are displayed here, though. This list is refreshed automatically. For detailed information you should use the main Call Log (p. 108).

**Event Log**
This list contains the same information as the main Event Log (p. 108). Only entries that have occurred since the last startup of TERAVoice are displayed here, though. This list is refreshed automatically. For detailed information you should use the main Event Log.

**CALL LOG**
This topic describes the Call Log. To display the Call Log expand the Monitor node on the tree-view pane. Click on the Call Log node to display the Call Log.

The Call Log displays all calls that have occurred. You can double click an item or right-click and select Properties in order to display details for the selected call.

**Call Detail Display**

**Navigating Call Log Records**
You can either select a different call in the Call Log list or you can use the arrow-buttons in order to navigate up and down the list.

**Copying Call Information**
You can click on the copy button in order to copy all information which is displayed.

**EVENT LOG**
This topic describes the Event Log. To display the Event Log expand the Monitor node on the tree-view pane. Click on the Event Log node to display the Event Log.

The Event Log displays all calls that have occurred. You can double click an item or right-click and select Properties in order to display details for the selected call.
Event Detail Display

Navigating Event Log Records
You can either select a different Event in the Event Log list or you can use the arrow-buttons in order to navigate up and down the list.

Copying Event Information
You can click on the copy button in order to copy all information which is displayed.

Online Help
For all errors and warnings you can click on the Online Information button to open a Web Browser with a link to the TERAVoice Support Database. We are constantly updating this database in order to provide help on frequently encountered errors.
TERAVoice Client Application

TERAVOICE CLIENT - WELCOME

Welcome and thank you for using TERAVoice. TERAVoice Client provides users with a quick access to their mailboxes. You can view current call logs and listen to new messages. TERAVoice Client runs in the background and pops up a notification window each time a new message or a new missed call is available.

- Installation (p. 110) gives instructions how to install TERAVoice Client and how you automate installation of TERAVoice Client on a group of computers
- Overview (p. 111) provides a quick reference of all functions and menu commands provided by TERAVoice Client
- Options (p. 112) describes how to set up TERAVoice Client to monitor certain mailboxes and to select the TERAVoice Server machine

TERAVOICE CLIENT INSTALLATION

Manual Installation

The setup package for the TERAVoice Client application is automatically installed into the following directory in case it is not deselected during TERAVoice Setup:

<Program Files Directory>\TERAVoice Server\TVShare\Client

The setup program automatically creates a hidden network share of the TVShare directory named ‘TERAVoice_0$’ during installation. With this method users can access the TERAVoice Client setup program by executing the link:

\\SERVERNAME\TERAVoice_0$\Client\Setup.exe

Where 'SERVERNAME' is the network name of the TERAVoice Server.

Tip: You can send this link to TERAVoice users in an e-mail to inform them how to install TERAVoice Client

Automated Installation

Windows Installer

The TERAVoice Client Setup is a Windows Installer package that can be used with the automatic software packaging feature of Windows 2000 and later.
The administrator can create a group policy that forces automatic installation for all users that this policy applies to.
Please refer to the documentation of Windows for detailed instructions how to create or modify a group policy for automatic software distribution.

**Systems Management Server**
The TERAVoice Client Setup also includes a PDF (package description file) file that can be used with MS Systems Management Server or other applications for automated software installation.

## TERAVOICE CLIENT OVERVIEW

### General
The TERAVoice Client application is automatically installed into the Startup programs folder and is automatically started after logon.
TERAVoice Client runs in the system tray indicated by the TERAVoice icon.
By right-clicking on this icon you can:
- Open the TERAVoice Call Log window
- Display the options (p. 112) dialog or
- Exit TERAVoice Client

### Call Log Window
The Call Log window displays a list of all calls that occurred for a certain mailbox. If you are monitoring more than one mailbox, tabs are displayed above the list for selecting the desired mailbox.

![TERAVoice Calls](image)

### Message Playback
When a message is available for a certain call this is indicated by a mail envelope icon and you can listen to the selected message by double-clicking the message, clicking on the playback button in the status bar or choosing 'Play' from the 'Tools' menu.

### Call Details
You can view details for a selected call by selecting 'Show Details' from the 'Tools' menu or by right-clicking a list item.
Copy List Items
If you would like to perform further processing of the call log, you can copy the selected list items to the clipboard by selecting 'Copy selected items' from the 'Edit' menu. To copy the entire list, you must choose 'Select all entries' first.

The selected entries are copied to the clipboard including column headers. Columns are separated by tab characters.

TERAVOICE CLIENT OPTIONS

Choose TERAVoice Server
Selecting options for TERAVoice Client is quite simple. First, you need to specify the name of the TERAVoice Server by clicking on the button behind the server name field.

If you are starting TERAVoice Client for the first time, the server selection dialog will be started automatically.

Selecting Mailboxes
If a connection to the TERAVoice server could be established successfully, you can choose the mailboxes you want to monitor by selecting the checkboxes.

Note: You need to have permissions to access a certain mailbox. If you do not have sufficient permission, you will see an empty list. Please contact your administrator for setting up permissions.

Poll Interval
The poll interval determines how often TERAVoice Client should check for any new calls on the selected mailboxes.
This chapter describes TERAVoice can be used to create custom IVR applications. TERAVoice supplies two different yet similar API’s that give you complete control about how calls are handled, input is processed and messages or TTS prompts are played back:

- **Scripting API**
- **COM API**

**SCRIPTING API**

The scripting API is the easiest way to develop custom IVR applications. Programming with this API is very similar to and as easy as creating ASP based web applications.

TERAVoice comes with an integrated Script Editor (p. 128) with syntax highlighting that can be used to edit the scripts. TERAVoice supports the following Scripting languages:

- VBScript
  - Microsoft Visual Basic Script or
- JScript
  - Microsoft’s implementation of JavaScript

You can use different scripting languages like Perl or Python but these languages are not officially supported.

How to create and use scripts is described in the next topic, Using Scripts.

**COM API**

If you prefer the robustness of compiled components and strong typing you can choose the COM API for developing IVR applications.

You can choose any programming language you like which is capable of creating polymorphic COM components (via IDispatch). Polymorphic in COM means that a component can have more than a single interface. Most COM enabled programming languages are able to do so. Examples are:

- Microsoft Visual C++
- Microsoft Visual Basic
- Borland Delphi
- etc.

Creating COM components for implementing custom IVR functionality with TERAVoice is as easy as creating the component in your preferred programming language and implementing a single interface that TERAVoice provides.
More information about creating COM components for use with TERAVoice is provided under **Using COM**.

**SUPPORT FOR .NET**

Though it might be possible to create a custom COM callable wrapper that implements the required interface, TERAVoice currently does not directly support .NET as a development platform.

**COMPARISON**

<table>
<thead>
<tr>
<th>Advantages</th>
<th>Disadvantages</th>
</tr>
</thead>
</table>
| **Scripting API** | Easy to implement  
Integrated script editing  
No compilation necessary  
No installation necessary  
Easy Debugging with Script Debugger | No compilation, no full syntax-checking  
performance penalty when doing excessive calculations |
| **COM API** | Strong typing  
Syntax check during compilation  
Language more powerful than Scripting language | Component must be installed on the TERAVoice server  
IDE must be installed on the TERAVoice server for debugging |
Scripting API

GENERAL

TERAVoice uses the scripting capabilities that are integrated into the Windows operating system and are used by the Windows Scripting Host but also for server-side scripting in ASP web-pages by IIS.

Scripting Languages

Two scripting languages are installed with Windows by default:

- VBScript
  Microsoft Visual Basic Script and
- JScript
  Microsoft’s implementation of JavaScript

Officially only these two languages are supported by TERAVoice but there should be no problem with using alternate scripting languages like Perl of Python.

IMPLEMENTATION

Events

For each Call Handler of the IVR Script type there is one script which you can edit with the integrated Script Editor (p. 128). The script must contain a set of functions that are called by TERAVoice when certain events occur. These functions are described under Events Reference (p. 135).

Note: If one of these functions is missing from your script, the script will fail and the call will be disconnected. It is recommended that you always either start from a sample script or select Insert Default Template from the Insert menu. The default template contains all the required functions.

You can add any number of additional functions to your script.

Properties and Functions

There are a number of Properties (p. 138) and Functions (p. 142) you can use to communicate with TERAVoice. All those properties and functions are members of the CallController object. With the scripting implementation this object is global for your script, so that you can always directly use all properties like global variables and all functions like global functions.

STEPS TO CREATE AN IVR SCRIPT

In this section we are going to lead you through the process of creating an IVR Script.
Create the Call Handler

Open the TERAVoice administration console and expand the Call Handlers node to select the IVR Modules node. Right-click and select New ⇒ IVR Module - Script.

Enter a name (e.g. Test) and press OK. The new item appears in the right pane. Right-click on it and select Properties.

Configuring the Call Handler

On the first property page clear the Use Default check-box and enter 0s. Then click on the Set as default button to be sure that a call is taken by this call handler when testing later. Of course you can also enter a Call Handler Assignment (p. 95) entry if you do not want to change the default Call Handler.

Now switch to the second tab named Script. Select VBScript as the script language and then click on the button to open the Script Editor (p. 128).

Creating the Script

You can now see the Script Editor with an empty script. Select Default Template from the Insert menu to insert the set of functions that need to be present in the script.

We are only going to create a very simple application in this section. More useful examples can be found under Scripting Samples (p. 131).

The goal of our Script is to play back a message, repeat the message if the caller presses '1' and hang up the call if the caller presses '0'.

The first step is playing an audio file when the Call Handler is started. Go to the function CallHandler_TimePulse and enter the following lines:

\[\text{If IsPlayed Then Exit Sub}\]
\[\text{PlayAudioFile SysAudioFolder \& \"\Help.wav\"}\]
\[\text{IsPlayed = True}\]

Since we need the IsPlayed variable to be global to our script we need to declare it outside of the function. Go to the top of the script and enter the following line after the Option Explicit statement:

\[\text{Dim IsPlayed}\]

The CallHandler_TimePulse function will be called in 1 second intervals and additionally every time when another event has occurred or the state of the call has changed. The function PlayAudioFile (p. 143) will playback the specified audio file. You can supply a complete path to any audio file you like here.

The SysAudioFolder points to the messages folder of TERAVoice's default language. The Help.wav is an audio file used by TERAVoice for help in remote control mode and is used here for testing purposes only.

Next we need to process the digits that the user presses. Go to the function CallHandler_DigitReceived and enter the following lines:

\[\text{Select Case Digit}\]
What happens here is very obvious. When the user presses '1' the same function is called that we have already used in the CallHandler_TimePulse event. When the user presses '0' the call is disconnected by calling the HangUp Function (p. 143).

**Testing the Script**

Press Save and Close to close the Script Editor. The script is automatically saved within TERAVoice. If you want to create a backup copy of your script or want to use it in another Call Handler you can use the Save as... and Open functions to load and save from and to script files.

Now you can use the Check Declaration of Functions button to check if you have implemented all the necessary functions and used the correct number of parameters etc.

**Please note:** This check does not test your code for any other errors. Since we are using scripting which is an interpreted language no compilation is being done. Any errors will not be detected before the actual execution of the code.

Now make sure that TERAVoice is running and make a call. You should now hear the help message being played. Try pressing '1' to start the message from the beginning and '0' to disconnect the call.

If you are unable to connect the call you should read the chapter on Troubleshooting (p. 147) for tips on how to solve the problem.

**Implementing Error Handling**

TERAVoice automatically reports errors that occur in your script and writes the errors to the Call Log (p. 108). Therefore it is not necessary for you to implement error handling in your script if you only want to get notified of the error. The following code is not necessary:

```vbscript
On Error Goto ErrHandler

....

Exit Function
ErrHandler:
    WriteCallLog err.Number, err.Description,...
```

Of course, for all other purposes, using error handling in your script could be quite useful.
FINDING ERRORS

If you notice that your script has failed or doesn’t work as expected, the first step always is having a look at the Call Log (p. 108). The details property page shows all events that have taken place and all script errors that have occurred. All errors are written to the Event Log (p. 108) as well but the entries in the Call Log are better suited for this purpose because they are always related to a certain call.

You can use the WriteCallLog function in your script to write entries to the Call Log. With this method you can easily keep track of how far processing has gone, which functions have been called and you can output the value of important variables.

If this doesn’t solve your problem and you cannot find the error you should consider debugging your script:

DEBUGGING SCRIPTS

TERAVoice supports debugging of all scripts you create. The following requirements need to be met in order to enable debugging:

- A Script Debugger must be installed
- Debugging of scripts must be enabled in TERAVoice

Script Debugger

It is very likely that you already have a script debugger installed. Several applications like MS Office, MS Visual Studio etc. install a script debugger. If you are unsure whether you have a script debugger installed you can download one from the Microsoft Web Site.

Enable Debugging

To enable debugging with TERAVoice you need to turn on script debugging on the Advanced Options (p. 94) property page. When debugging is enabled the timeout value is disabled automatically.

**Important note:** When debugging is enabled all script execution is marshaled to a single thread. TERAVoice is a multi-threaded application and uses a separate thread for each call, but when script debugging is turned on, scripts that are running simultaneously can block each other! Therefore you should turn on script debugging only for testing purposes and not on a production system.

Debug Scripts

Once you have enabled debugging the debugger should automatically appear every time an error occurs in your script.

To open the Script Debugger from the beginning you can insert the following line as the first statement in the CallHandler_Init function.

```plaintext
Stop
```

The debugger will open as soon as the statement is reached. You need to manually set an execution to the next statement to continue.
**Tip:** If you want to stop debugging and have the script continue without the debugger it is best to choose the option Detach instead of Stop Debugging.
COM API

GENERAL

In addition to the Scripting API (p. 117) you can implement custom IVR behavior by creating COM (ActiveX) components that TERAVoice creates when you select an IVR Module Call Handler (p. 101) of the COM Component type.

The advantages and disadvantages of this approach are described under Developing IVR Applications (p. 115).

IMPLEMENTATION

Each COM component usually has its own set of methods/functions and properties which are exposed via the default automation interface of the component. If you were to create a usual custom COM component TERAVoice would not be able to use it because it doesn’t know anything about this component and its interfaces.

Therefore your component needs to implement an interface that TERAVoice already knows about. This interface is called ITVCallHandler and is provided as a Type Library in the TERAVoiceLib.dll file. This file and its Type Library are automatically registered during TERAVoice installation.

To enable your component to be used with TERAVoice, your COM component needs to implement this interface. The interface defines several functions that you need to implement. TERAVoice calls these functions in your object then when certain Events (p. 135) occur.

Interface Details

- ProgID: TERAVoiceLib.ITVCallHandler
- TypeLib ID: {C6E3BA98-2C72-4738-8AD1-B605C1BB1F62}
- TypeLib Description: TERAVoice Extensions Library
- Defined in: TERAVoiceLib.dll

This Type Library also defines the TERAVoiceLib.CallController object that you can use to get information from TERAVoice (Properties (p. 138)) and control TERAVoice (Functions (p. 142)).

Requirements

The component you create should be an in-process component in a DLL (dynamic link library). Though it might be possible to create an out-of-process component (ActiveX server), there are several reasons against doing so:

- Performance would suffer because marshalling to a different process (the process of the ActiveX server) is required
- If you declare the COM component as Multi-Use, several Call Handlers that are using the same COM object and are running simultaneously could block each other because they would run in the same thread
You could prevent this by declaring the COM component as single use but then the only advantage of an out-of-process component would be gone: You cannot use global variables to communicate between these objects.

These reasons especially apply to Visual Basic development, because VB doesn’t support creating multi-threaded components.

**Synchronization**

If you want to synchronize several COM Component Call Handlers that are running simultaneously the best ways are:

- Storing synchronization data in the registry, in the file or inside a database
- Have your in-process Call Handlers communicate with an out-of-process COM Component (ActiveX Server) that does not have any functionality built in that could block execution in any way (best case would be a component that does not have any methods, but just properties).

**STEPS TO CREATE THE COM COMPONENT**

While you can use any programming language that supports creating COM components, the following paragraph describes the process for creating such a component for MS Visual Basic and MS Visual C++ only. If you are familiar with creating COM components with a different programming environment, it should be easy to find the right steps for your preferred development environment, though.

**MS Visual Basic**

- Create a new project of type 'ActiveX DLL'
- Rename your project from 'Project1' (like 'MyProject')
- Rename 'Class1' to something else (like 'Test1')
- Select *Project > References* and select the entry 'TERAVoice Extensions Library'
- Open the code window for your class and enter the following lines:

  ```vbnet
  Implements ITVCallHandler
  Dim MyCallController as CallController
  ```

- Select ITVCallHandler from the left combo box, then select each item in the right combo box until all interface functions are present
- Go to the ITVCallHandler_Init function and enter:

  ```vbnet
  Set MyCallController = objCallController
  ```

- Go to the ITVCallHandler_Init function and enter:

  ```vbnet
  Set MyCallController = Nothing
  ```
Now you can start adding your code

You can also have a look at the Visual Basic Samples (p. 132) or start by copying and renaming one of the samples.

**Note:** You do not need to use Binary Compatibility for your COM components. If you do not use Binary Compatibility, new GUIDs (COM registration information in the registry) are created every time you compile your project. Since TERAVoice only uses the ProgID of your component to create the object and the ITVCallHandler interface (which never changes) to access the object, you don't need to care about component compatibility.

**MS Visual C++ 6.0**

- Create a new project and select the ATL COM AppWizard
- Select the desired folder and enter a name for the project. This name will be the left part of your component’s ProgID
- Select Dynamic Link Library and press OK
- Right-click the root node in the ClassView and select New ATL Object
- Select simple object and press Next
- Enter the name of your object as the Short Name, this will be the right part of your component's ProgID
- Accept all defaults by pressing OK
- A new class appears in ClassView with the chosen name preceded by a 'C'
- Right-click on this item and select Implement Interface
- Scroll down to select 'TERAVoice Extensions Library', then press OK
- Select the ITVCallHandler interface and press OK
- The implementation of all required functions is now added to the header file
- Now you need to add all functions you want to use to your class. For this purpose you need to prefix the functions to avoid naming conflicts. Example:

  Right-click the class and select Add member function, enter **long** in the first line and in the second line: **m_DigitReceived(BSTR *Digit)** and press OK

  Double-click on the DigitReceived function and replace the line

  ```cpp
  return E_NOTIMPL;
  ```

  with

  ```cpp
  return m_DigitReceived(Digit);
  ```
If you do not want to use all functions you should change

```c
return E_NOTIMPL;
```

to

```c
return S_OK;
```

for all those functions to avoid errors during runtime

During the `Init` (p. 136) function you should store a reference to the `CallController` object (call `AddRef`) and during `UnInit` (p. 137) you need to release the object.

Now you can start adding code to your functions

**MS Visual C++.NET 7**

- Create a new project and select *C++ Projects* ⇒ *ATL Project*

- Select the desired folder and enter a name for the project. This name will be the left part of your component's ProgID

- Press finish on the ATL Project Wizard to accept the default settings

- Open the Solution Explorer and remove the Proxy Stub project. This is the second project with the project name followed by 'PS'. If you use this COM object locally only you do not need it.

- Right-click the root node in the ClassView and select *New* ⇒ *Class*

- Select ATL Simple Object and press Open

- Enter the name of your object as the *Short Name*; this will be the right part of your component's ProgID. Press *Finish*.

- A new class appears in ClassView with the chosen name preceded by a 'C'

- Right-click on this item and select *Add* ⇒ *Implement Interface*

- Select 'TERAVoice Extensions Library' from the dropdown list.

- Select the `_ITVCallHandler` interface and press on the arrow button to add it to the list on the right. Press *Finish*.

- The implementation of all required functions is no added to the header file
Now you need to add all functions you want to use to your class. For this purpose you need to prefix the functions to avoid naming conflicts. Example:

Right-click the class and select Add ⇒ Add function, enter long as Return type, private as Access type and m_DigitReceived as the Function name. Enter BSTR * in the Parameter type field and Digit as Parameter name, then press Add. Click on the button near the .cpp file box and select the file C<YourClassName>.cpp - then press finish.

Double-click on the DigitReceived function and replace the line

```cpp
return E_NOTIMPL;
```

with

```cpp
return m_DigitReceived(Digit);
```

If you do not want to use all functions you should change

```cpp
return E_NOTIMPL;
```

to

```cpp
return S_OK;
```

for all those functions to avoid errors during runtime

During the Init (p. 136) function you should store a reference to the CallController object (call AddRef) and during UnInit (p. 137) you need to release the object.

Now you can start adding code to your functions

**FINDING ERRORS**

If you notice that your component has generated an error or doesn’t work as expected, the first step always is to have a look at the Call Log (p. 108). The details property page shows all events that have taken place and all script errors that have occurred. All errors are written to the Event Log (p. 108) as well but the entries in the Call Log are better suited for this purpose because they are always related to a certain call.

You can use the CallController::WriteCallLog function in your component to write entries to the Call Log. With this method you can easily keep track on how far processing has gone, which functions have been called and you can output the value of important variables.

If this doesn’t solve your problem and you cannot find the error you should consider debugging your component:

**DEBUGGING COMPONENTS**

How to debug your COM components depends on your development environment. Please refer to the documentation of your development environment on how to debug COM component. Some notes about MS Visual Basic and C++ can be fond in the following sections:
**Important:** The TERAVoice service is running under the LocalSystem account by default. In some situations it might be necessary to change the Logon information of the TERAVoice service to the current user in order to allow debugging to work properly.

### MS Visual Basic

Debugging your component with Visual Basic is as easy as starting to debug (with option 'Wait for components being created) and setting your breakpoints.

When debugging a DLL with Visual Basic your component somehow behaves like an out-of-process component (ActiveX server) because all component calls are marshaled to the Visual Basic Debugger which runs in its own process. The same restrictions apply as noted above. Of course you should not use debugging in production mode and avoid several simultaneous calls (that are handled by a COM object that is served by the same VB project in debug mode).

**Note:** If you can't avoid this situation this might run without major problems if you do not have any large processing or blocking functions in your project and do not switch to break mode for too long a time. In that case OLE timeouts could occur which could cause TERAVoice to fail!

You can run several instances of Visual Basic though to debug several different VB projects simultaneously with no problem.

### MS Visual C++

The debugging model with Visual C++ is completely different. There are no functions to marshal calls to a debugger running out of process. The only option is to natively debug the whole TERAVoice process.

To debug with Visual C++, load your project, switch to the Debug configuration and build your project. Set the desired breakpoints in your project now.

Select Debug ⇒ Processes and locate the TERAVoice.exe process in the list and select Attach. Now call into TERAVoice and wait until one of the breakpoints is reached.
Script Editor

The Script Editor can be used for editing scripts. It offers Syntax-Highlighting, code checking and all common editor functions.

The Script Editor is started by opening the property page of an IVR Module Call Handler (p. 101) of the script type and clicking on the 'Open Script Editor' button or directly via right-click on the Call Handler.

When you open the Script Editor from the Administration Console you always edit the selected script which is automatically saved to TERAVoice when you close the Script Editor.

Alternatively you can directly open the Script Editor with its Start Menu entry to edit script files that are not stored within TERAVoice.

MENUS

File Menu
With the 'New' command your current script is discarded and a blank script is displayed.
Use the 'Open' and 'Save' commands to load and save script files.
The 'Print' command lets you print your script or the current selection.

Edit Menu
The commands in the edit menu are common to generic Windows programs and behave as expected.
From the 'Find' dialog you can additionally choose to mark all selections by setting bookmarks.

View Menu
Use the 'View' menu to show or hide toolbars or the status bar.

Tools Menu
With the 'Insert Default Template' command you can insert a template that implements all the required functions for the currently selected scripting language. This feature only works with VBScript and JScript.
The 'Check Code' commands checks your current script for the correct implementation of all required functions and performs a basic syntax check on your script.
Use the 'Options' command to display a dialog that lets you configure various settings to customize the editor behavior.

TOOLBARS

Standard Toolbar
The standard toolbar contains frequently used commands that are present in the menus described above.
Bookmarks Toolbar

Use the bookmarks toolbar to manage bookmarks in your script. By using bookmarks you can easily mark positions of interest and jump between these.

- 🔖 Toggle bookmark on current line
- 🔖 Jump to next bookmark
- 🔖 Jump to previous bookmark
- 🔖 Clear all bookmarks

You can use the 'Find' command to automatically create a bookmark on all lines containing a search string.
Creating Outbound Calls

You can create outbound calls with TERAVoice by using the feature for scheduled IVR Call Handler execution.

Please refer to the topic IVR Module Call Handler (p. 101) for details on how to activate scheduled execution of IVR Call Handlers.

When your IVR Module is executed from a scheduled execution you don't receive the `CallHandler_Init` event but the `CallHandler_ScheduledInit` event. You can use this event to perform any maintenance tasks or you can call the `CreateCall` function during this event.

If the outgoing call fails you receive the `CallHandler_OutgoingCallFailed` event, in case of success you will get the normal `CallHandler_Init` event and the call can be processed as usual.

For an outbound call example please refer to the WakeupCall sample application.
IVR Samples

SCRIPTING SAMPLES
TERAVoice comes with samples for both supported scripting languages, VBScript and JScript.

VBScript
There are two VBScript sample scripts. You can find these scripts in the folder:

<TERAVoice_Installation_Folder>\Samples\VBScript

For further samples you can also have a look at the Visual Basic Samples (p. 132). This code can be easily ported to VBScript by replacing type declaration and changing some other things which are not supported in VBScript.

Calculator
The Calculator sample Application implements a simple calculator behavior. You can enter two numbers and the sum of those numbers is announced to the caller:

- Enter a number of one or more digits
- Press the '*' key
- Enter a second number of one or more digits
- Press the '#' key
- The result of the calculation will be announced as a sequence of digits

Important note: In order to get the script application to work you need to change the BasePath variable specified in one of the first lines of the script depending on the path you have chosen to install TERAvoice. The path must point to the <TERAVoice_Installation_Folder>\Samples\Visual Basic 6\Calculator folder, because this folder contains the required audio files.

Calculator TTS - Text to Speech
The second VBScript sample is very similar to the Calculator sample except that it uses Text-to-Speech for the voice prompts.

Tip: Be sure that you have an English TTS engine selected in the Server Options - Advanced Settings (p. 94) dialog or change the text strings in the script to the language of your selected TTS engine.

JScript
A JScript sample is included in the following folder:
<TERAVoice_Installation_Folder>\Samples\JScript

**Calculator**
This is the same sample like the VBScript Calculator which was translated to JScript. Please note the difference in building the array for the PlayAudioFileList (p. 143) function. This is due to the fact that JScript doesn’t support VBScript Safe Arrays.

Don't forget to adjust the BasePath variable as described above.

**VISUAL BASIC SAMPLES**
TERAVoice comes with three samples for Visual Basic 6. None of these samples is using functionality that is specific to Visual Basic 6, so they could easily be ported back to Visual Basic 5 if required.

You can find the Visual Basic samples in the following folder:

<TERAVoice_Installation_Folder>\Samples\Visual Basic 6

**Calculator**
The Calculator sample Application implements a simple calculator behavior. You can enter two numbers and the sum of those numbers is announced to the caller:

- Enter a number of one or more digits
- Press the '*' key
- Enter a second number of one or more digits
- Press the '#' key
- The result of the calculation will be announced as a sequence of digits

To test this sample, enter ‘tvSample1.Calculator’ as ProgID in the IVR Module Call Handler (p. 101) configuration dialog.

**Features**
This sample demonstrates some of the common features and functions that can be used with TERAVoice. You can also use this sample if you want to test if your installation supports certain features.

After the call is connected a message is played back, telling you about the available features you can select by pressing a key on the telephone:

- '0' - Play welcome message again
- '1' - Switch to different Call Handler by entering the ID
- '2' - Put call on hold
- '3' - Record 10s of audio and play back afterwards
- '4' - Transfer call to a phone number you can enter
'5' - Play the CallerID
'6' - Hang up the call

To test this sample, enter 'tvSample2.Features' as ProgID in the IVR Module Call Handler (p. 101) configuration dialog.

Call Distribution
This sample demonstrates how to distribute calls to certain employees depending on the CallerID. This is done by looking up the CallerID in a database (a sample database is provided).

As soon as a call is received the CallerID is looked up in the table 'Customers'
If a match is found the AssistantID and the phone number of the assistant is looked up
If no match is found a default phone number (e.g. operator) is used
The call is then transferred to the resulting destination
If the transfer fails a message is played back

Of course this sample requires some enhancements for a production use. You can implement subsequent transfer targets which can be used if transfer fails (or the transfer target does not answer).

Please note: This sample will only work if your telephony hardware and your PBX or public network support CallerID (p. 138) and Call Transfer (p. 35).

To test this sample, enter 'tvSample3.CallDistribution' as ProgID in the IVR Module Call Handler (p. 101) configuration dialog.

WakeUp System
This is a demonstration of outbound call creation. To use this application you must set up the COM Call Handler to use scheduled execution as described in topic IVR Module Call Handler.

- The sample comes with a database that contains a list of wake-up calls to perform each on a certain date and time
- Upon scheduled execution the application receives the CallHandler_ScheduledInit event. In this event it checks the wake-up call database for any calls that might be due.
- In case a due call is found it calls the CreateCall function and exits the event procedure in order to wait either for the CallHandler_OutgoingCallFailed event or for the CallHandler_Init event which would mean that the call has been taken by the remote party
- The number of attempts are logged and the call is marked as succeeded or failed depending on the results

Note: This sample (and outgoing call applications in general) will not work with normal voice modems because voice modems are unable to report if an outgoing call has been answered or not!
To test this sample enter ‘tvSample4.Wakeup as ProgID in the IVR Module Call Handler (p. 101) configuration dialog and do not forget to activate scheduled execution.

C++ SAMPLES

TERAVoice comes with one sample for MS Visual C++ that is available for both Versions MS VC 6.0 and MS VC 7.x (MS VC.NET, not using managed extensions).

Calculator

The Calculator sample Application implements a simple calculator behavior. You can enter two numbers and the sum of those numbers is announced to the caller:

- Enter a number of one or more digits
- Press the "*" key
- Enter a second number of one or more digits
- Press the "#" key
- The result of the calculation will be announced as a sequence of digits

Visual C++ 6.0

You can find the sample for MS Visual C++ 6.0 in this folder:

<TERAVoice_Installation_Folder>\Samples\Visual Basic 6 C++ 6.0\tvC6Sample1

To test this sample, enter ‘tvC6Sample1.Calculator’ as ProgID in the IVR Module Call Handler (p. 101) configuration dialog.

Visual C++ 7.0

You can find the sample for MS Visual C++ 7.x in this folder:

<TERAVoice_Installation_Folder>\Samples\Visual Basic 6 C++ 6.0\tvC7Sample1

To test this sample, enter ‘tvC7Sample1.Calculator’ as ProgID in the IVR Module Call Handler (p. 101) configuration dialog.

Note: You need to change the BasePath setting in the constructor of the Calculator class in a way that it points to the folder of the Visual Basic calculator sample because the required audio files are located in this folder.
IVR API Reference

EVENTS

The events described here are not real COM events but events from a logical point of view. How the events are actually implemented depends on whether you are using Scripting or the COM API.

When you are using Scripting you have to simply implement all of these events as functions that are being called by TERAVoice.

With the COM API you need to implement the interface ITVCallHandler which is defined in the TERAVoiceLib type library (friendly name: TERAVoice Extensions Library). The ITVCallHandler interface defines all those functions that get called by TERAVoice as soon as an event occurs.

**CallLogEntryEvent**

**Definition:**

```
CallHandler_CallLogEntryEvent(EntryID, CallDate, CallerID, CallHandlerID, CallHandlerName, CallHandlerType, CallDetails, CallDetailsAddition, MessageLength, MessageFileName)
```

**Parameters:**

- `EntryID` (string): ID of the call
- `CallDate` (datetime): date and time of the call
- `CallerID` (string): phone number of the caller
- `CallHandlerID` (long): ID of the last Call Handler that was active
- `CallHandlerName` (string): name of the last active Call Handler
- `CallHandlerType` (string): letter depending on the type of Call Handler (see Call Handler types (p. 138))
- `CallDetails` (string): contains all of the detail information separated with line breaks
- `CallDetailsAddition` (string): you can add information to the CallDetails by writing into this variable (all other parameters cannot be changed)
- `MessageLength` (long): length of a recorded message, 0 if none was recorded
- `MessageFileName` (string): file name of the recorded message

**Description:**

This function is called each time an entry is added to the Call Log.

No outbound call can be created during this event though. To create an outbound call you need to add an entry to a database or any other storage medium and perform the outbound call during scheduled execution of an IVR Call Handler.
### CallTransferResult

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandler_CallTransferResult(Succeeded)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters:</td>
<td>Succeeded (boolean): true when the transfer was successful, false if not</td>
</tr>
<tr>
<td>Description:</td>
<td>This function is called after an attempt to transfer a call has either succeeded or failed. When the transfer has failed you can try to start another transfer to a different destination. To transfer a call you need to call the TransferCall (p. 145) function.</td>
</tr>
</tbody>
</table>

### DigitReceived

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandler_DigitReceived(Digit)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters:</td>
<td>Digit (string): contains the digit that was pressed by the user</td>
</tr>
<tr>
<td>Description:</td>
<td>This function is called every time TERAVoice detects a caller pressing a key on the touch tone keypad of his telephone.</td>
</tr>
</tbody>
</table>

### FileSaved

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandler_FileSaved(FileName, Length)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters:</td>
<td>FileName (string): filename and path of the audio file</td>
</tr>
<tr>
<td></td>
<td>Length (long): length of the recorded audio file in seconds</td>
</tr>
<tr>
<td>Description:</td>
<td>This function is called when recording of an audio message is completed.</td>
</tr>
</tbody>
</table>

### Init

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandler_Init(objCallController)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters:</td>
<td>none when using Scripting</td>
</tr>
<tr>
<td></td>
<td>objCallController (COM object) when using the COM API</td>
</tr>
<tr>
<td>Description:</td>
<td>This function is the first function that is called when the IVR Call Handler is loaded. This function is only called once. You can use this function to initialize your application. When using the COM API you should store the CallController object in a local variable. You need this object to execute functions.</td>
</tr>
</tbody>
</table>
### OutgoingCallFailed

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandler_OutgoingCallFailed()</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters:</td>
<td>none</td>
</tr>
<tr>
<td>Description:</td>
<td>This function is called when an attempt to create an outgoing call has failed or the call was not taken within the requested period of time</td>
</tr>
</tbody>
</table>

### PlayDone

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandler_PlayDone()</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters:</td>
<td>none</td>
</tr>
<tr>
<td>Description:</td>
<td>This function is called when an audio file has finished playing.</td>
</tr>
</tbody>
</table>

### ScheduledInit

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandler_ScheduledInit(objCallController, InstanceCount)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters:</td>
<td>objCallController (COM object) when using the COM API</td>
</tr>
<tr>
<td></td>
<td>InstanceCount: number of scheduled instances of this Call Handler currently running</td>
</tr>
<tr>
<td>Description:</td>
<td>This function is called when an IVR COM object or script is executed as of scheduled execution.</td>
</tr>
</tbody>
</table>

### TimePulse

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandler_TimePulse()</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters:</td>
<td>none</td>
</tr>
<tr>
<td>Description:</td>
<td>TERAVoice calls this function in intervals of 1 second and additionally every time some status has changed or another event has occurred (like PlayDone).</td>
</tr>
</tbody>
</table>

### UnInit

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandler_UnInit()</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters:</td>
<td>none</td>
</tr>
<tr>
<td>Description:</td>
<td>This function is called before the Call Handler gets unloaded. You can use this function do clean up you objects and uninitialize your application.</td>
</tr>
</tbody>
</table>
Don’t forget to release the reference to the CallController object during this event when you are using the COM API!

**Call Handler types**

The Call Handler type is determined by the following letters:

<table>
<thead>
<tr>
<th>Letter</th>
<th>Call Handler Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>M</td>
<td>User Mailbox</td>
</tr>
<tr>
<td>S</td>
<td>Voice Menu</td>
</tr>
<tr>
<td>T</td>
<td>Time Schedule</td>
</tr>
<tr>
<td>R</td>
<td>Remote Control</td>
</tr>
<tr>
<td>I</td>
<td>IVR Module - Script</td>
</tr>
<tr>
<td>Q</td>
<td>Waiting Queue</td>
</tr>
<tr>
<td>H</td>
<td>Music On Hold</td>
</tr>
<tr>
<td>A</td>
<td>IVR Module - COM</td>
</tr>
<tr>
<td>N</td>
<td>H.323 User</td>
</tr>
</tbody>
</table>

**PROPERTIES**

The properties described in this topic supply information about the current call. If you are using the Scripting API (p. 117) you can directly use these properties as variables in your code. When using the COM API (p. 122) you can access these as properties of the CallController object which is supplied to your application during the ITVCallHandler_Init event.

*Note: All properties are read-only.*

**CallerID**

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallerID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type:</td>
<td>string</td>
</tr>
<tr>
<td>Description:</td>
<td>If you telephony hardware supports reporting CallerID this property contains the number of the calling party. If no CallerID was transmitted the property is empty.</td>
</tr>
</tbody>
</table>
### CallHandlerDescription

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandlerDescription</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type:</td>
<td>string</td>
</tr>
<tr>
<td>Description:</td>
<td>Contains the description of the IVR Call Handler as entered on the general tab of the Call Handler property sheet.</td>
</tr>
</tbody>
</table>

### CallHandlerID

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandlerID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type:</td>
<td>long</td>
</tr>
<tr>
<td>Description:</td>
<td>The ID of the IVR Call Handler.</td>
</tr>
</tbody>
</table>

### CallHandlerName

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallHandlerName</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type:</td>
<td>string</td>
</tr>
<tr>
<td>Description:</td>
<td>Contains the name of the IVR Call Handler as entered on the general tab of the Call Handler property sheet.</td>
</tr>
</tbody>
</table>

### CallID

<table>
<thead>
<tr>
<th>Definition:</th>
<th>CallID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type:</td>
<td>string</td>
</tr>
<tr>
<td>Description:</td>
<td>TERAVoice generates a unique ID for every call that is taken. You can use this ID to uniquely identify a call.</td>
</tr>
</tbody>
</table>

### ConnectedID

<table>
<thead>
<tr>
<th>Definition:</th>
<th>ConnectedID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type:</td>
<td>string</td>
</tr>
<tr>
<td>Description:</td>
<td>Usually the same as the CallerID (p. 138). If a call gets transferred the connected ID may change to reflect the new partner (if your hardware and telephony network support that feature).</td>
</tr>
</tbody>
</table>

### DeviceGroupList

<table>
<thead>
<tr>
<th>Definition:</th>
<th>DeviceGroupList</th>
</tr>
</thead>
<tbody>
<tr>
<td>Property</td>
<td>Definition</td>
</tr>
<tr>
<td>-------------------</td>
<td>------------</td>
</tr>
<tr>
<td>DeviceList</td>
<td></td>
</tr>
<tr>
<td>DeviceName</td>
<td></td>
</tr>
<tr>
<td>DTMFRoutingResult</td>
<td></td>
</tr>
<tr>
<td>IsOnHold</td>
<td></td>
</tr>
<tr>
<td>IsPlaying</td>
<td></td>
</tr>
<tr>
<td>Property</td>
<td>Definition</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------------------</td>
</tr>
<tr>
<td>IsRecording</td>
<td><strong>IsRecording</strong></td>
</tr>
<tr>
<td>OtherPartyNumber</td>
<td><strong>OtherPartyNumber</strong></td>
</tr>
<tr>
<td>RedirectingID</td>
<td><strong>RedirectingID</strong></td>
</tr>
<tr>
<td>RedirectionID</td>
<td><strong>RedirectionID</strong></td>
</tr>
<tr>
<td>SysAudioFolder</td>
<td><strong>SysAudioFolder</strong></td>
</tr>
</tbody>
</table>
The functions described in this topic can be used to control TERAVoice in regard to the currently connected call. If you are using the Scripting API (p. 117) you can directly use these functions like global functions in your code. When using the COM API (p. 122) you can access these as methods of the CallController object which is supplied to your application during the ITVCallHandler_Init event.

### CreateCall

**Definition:** 
CreateCall(TargetNumber, Device, Timeout)

**Return value:** boolean

**Parameters:**
- TargetNumber (string): the phone number to call
- Device (string): the name of the device or device group that should be used to create the call
- Timeout (long): time in seconds to wait for the remote party to answer the call

**Description:**
This function can only be called during the ScheduledInit (p. 137) event!
You can pass either a device name or the name of a device group for the device parameter. This name can be obtained from the DeviceList (p. 140) and DeviceGroupList (p. 139) properties.
If this function succeeds it returns true otherwise false. A return value of true means that the call could be successfully created on one of the devices.
If the call is answered by the remote party the Init (p. 136) event is fired as for an inbound call.
If the call fails to connect (e.g. because the remote party is busy) or the call was not answered within the timeout period, the OutgoingCallFailed (p. 137) event is fired.

### GotoCallHandler

**Definition:** 
GotoCallHandler(CallHandlerID)

**Return value:** boolean

**Parameters:**
- CallHandlerID (long): the ID of the target Call Handler

**Description:**
This function hands over the processing of the current call to a different Call Handler.
If this function succeeds it returns true and the
<table>
<thead>
<tr>
<th>Method</th>
<th>Definition</th>
<th>Return value</th>
<th>Parameters</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>HangUp</strong></td>
<td><strong>HangUp()</strong></td>
<td>none</td>
<td>none</td>
<td>This function disconnects the current call. Playback and recording is stopped and the <code>UnInit</code> (p. 137) event is fired.</td>
</tr>
<tr>
<td><strong>HoldCall</strong></td>
<td><strong>HoldCall()</strong></td>
<td>boolean</td>
<td>none</td>
<td>This function tries to put a call on hold. This function returns <code>true</code> if the call was successfully put on hold. If the call was on hold before or the telephony hardware or network is not capable of holding calls, the function returns <code>false</code>.</td>
</tr>
<tr>
<td><strong>PlayAudioFile</strong></td>
<td><strong>PlayAudioFile(FileName)</strong></td>
<td>boolean</td>
<td>FileName</td>
<td>This function plays back an audio file over the telephone line. If there is currently another playback or recording in progress it is stopped. The function returns <code>true</code> on success, <code>false</code> if an error has occurred.</td>
</tr>
<tr>
<td><strong>PlayAudioFileList</strong></td>
<td><strong>PlayAudioFileList(Files())</strong></td>
<td>boolean</td>
<td>Files</td>
<td>This is similar to PlayAudioFile. The difference is</td>
</tr>
</tbody>
</table>
that you can supply a list of audio files here.
The reason for this function is that starting a playback takes some time and resources. When you need to play several short audio files in a row you can use this function to avoid gaps during playback.

All specified audio files need to be of the same format!

### PlayTTS

**Definition:**

PlayTTS(Text,[Speed])

**Return value:**

boolean

**Parameters:**

Text (string): the message that is to be synthesized

Speed (long, optional): adjusts the speed of the message (use values from -10 to +10, default is 0)

**Description:**

This function allows synthesizing TTS prompts using the default voice specified in Server Options (p. 91) or set by the SetTTSVoice (p. 145) function.

You can control playback speed by using the optional speed parameter.

If playback has successfully started, the function returns true, otherwise false.

### RecordAudioFile

**Definition:**

RecordAudioFile(FileName, ReplaceExisting, MaxSeconds)

**Return value:**

boolean

**Parameters:**

FileName (string): the full path and filename for the recorded audio file

ReplaceExisting (boolean): if true replaces an existing file of the same name; if false the function returns false if the file already exists

MaxSeconds (long): the maximum duration of the recording

**Description:**

This function stops any playback, TTS or recording operation and starts recording to the specified file name.

Returns true, if the function succeeds, otherwise false.
SetTTSVoice

**Definition:**  
SetTTSVoice(VoiceName)

**Return value:**  
boolean

**Parameters:**  
VoiceName (string): the name of the TTS voice that should be used for subsequent calls to PlayTTS (p. 144)

**Description:**  
This function can be used to change the voice for TTS synthesis. The value supplied for VoiceName must be one of the available TTS voices as in the Advanced Options (p. 94) selection box. If the TTS voice was successfully set to the specified voice name the function returns true.

StopPlayRecord

**Definition:**  
StopPlayRecord()

**Return value:**  
none

**Parameters:**  
none

**Description:**  
This function immediately stops any current playback, TTS or record operation.

TransferCall

**Definition:**  
TransferCall(TargetNumber, TransferMode, TransferTimeout)

**Return value:**  
boolean

**Parameters:**  
TargetNumber (string): the phone number to which the call should be transferred  
TransferMode (long): specifies the transfer mode  
TransferTimeout (long, optional): specifies the amount of time to wait for the remote party to accept the call. If not specified, the default value from the Call Transfer settings is used

**Modes:**  
0: Basic Transfer (no wait for answer)  
1: Basic Transfer (wait for answer)  
2: TAPI Blind Transfer (not recommended!)  
3: Create Conference  
4: Simulate Transfer (Software bridging)

**Description:**  
This tries to transfer the call to the target phone number using the specified mode. If the transfer was completed successfully, the function returns true. This does not mean that the call was successfully transferred so. You
need to wait for the CallTransferResult (p. 136) event to obtain information about the result of the transfer

<table>
<thead>
<tr>
<th>UnHoldCall</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Definition:</strong></td>
</tr>
<tr>
<td><strong>Return value:</strong></td>
</tr>
<tr>
<td><strong>Parameters:</strong></td>
</tr>
<tr>
<td><strong>Description:</strong></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>WriteLogFile</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Definition:</strong></td>
</tr>
<tr>
<td><strong>Return value:</strong></td>
</tr>
<tr>
<td><strong>Parameters:</strong></td>
</tr>
<tr>
<td><strong>Description:</strong></td>
</tr>
</tbody>
</table>
Troubleshooting and Support

5 TROUBLESHOOTING AND SUPPORT

Troubleshooting

HOW TO IDENTIFY ERRORS

Common Problems
The next topic, Common Problems (p. 148), contains information about the problems most commonly encountered. Please read through the available topics to see if can find a solution for your particular problem.

Checklist
Due to TERAVoice’s support for a broad range of hardware and telephony network types it is very likely that you will encounter problems or errors at some time when you try to activate or use a certain feature. When you experience problems, such as a selected feature or function not working as expected or a call being disconnected when it was expected that a certain function should be executed you should always check the following:

- **Does my telephony network support the desired functionality?**
  Ask your provider about support for a certain feature. Some features require explicit subscription in some networks.

- **Does my PBX (if any) support the desired functionality?**
  Please refer to the documentation of you PBX for more details. Often it is required that you enable certain features on the PBX ports which are connected to TERAVoice.

- **Does my telephony hardware support the desired functionality?**
  Please refer to the documentation that comes with your hardware.

- **Does the driver (TAPI or CAPI) support the desired functionality?**
  You can use the DeviceCapabilities button to check some of the supported features.

- **Did you configure the feature properly?**
  Please refer to the related topic in this documentation for configuration instructions.

Description of certain features and their requirements and limitation can be found under Device Features (p. 34).
**Event Log and Call Log**

In most cases you can find useful information in the Event Log (p. 108) and in the Call Log (p. 108). If a certain problem occurs the first thing you should do is look into the Event Log. Especially watch for entries of the error type.

Read the log entry and see if the information is useful in locating the problem. If the error message does not contain information that seems useful you can try the ‘Online Information’ button to open the TERAVoice support pages containing extended error information.

If you can't find an answer to your problem here you should consider making use of our other Online Support (p. 152) options.

**COMMON PROBLEMS**

**Activation fails**

The activation process is self-explanatory and usually errors are displayed with meaningful descriptions. There are a few things to consider for both types of activation (p. 22), though.

**Online Activation**

Online activation is done by connecting to the TERASENS activation server via the internet. In order to get this activation method working, you need to have an open connection to the internet. TERAVoice connects to the activation server via http protocol on port 80 like any web browser connects to a web site.

TERAVoice uses the proxy settings that are configured in the Internet Explorer options dialog. If activation fails and TERAVoice indicated that it is unable to connect, you should first check if you can open a web site using the Internet Explorer. If this is not possible you should check your proxy settings and your internet connection.

If online activation fails, you should try activating via email instead.

**Email Activation**

If you are trying to activate via email you should usually get a response within 5 minutes.

```
Note: Depending on refresh or poll intervals of your or your provider’s email system you might experience longer durations until you receive a response.
```

If you do not get any reply within 24h please contact TERASENS support (p. 152).

**Check your telephony hardware**

If you are unsure whether your telephony hardware is properly installed and working, you should first test your hardware with one of the methods described here, depending on you hardware's driver.

**TAPI Based Hardware**

The best way for testing your TAPI based telephony hardware is using the Windows Phone Dialer. This application is installed by default on all operating systems that are supported by TERAVoice. It is contained in the Start menu of Windows 2000 only. The following method for starting the Windows Phone Dialer works on all operating systems:

- Click on Start ➔ Run
- Enter `dialer.exe` and press OK
The Windows Phone Dialer opens
Select Edit → Options
Select Phone as the Preferred Line For Calling
In the dropdown list besides Phone Calls select your device from the list of available devices

**Note:** If your device is not listed here it means that the TAPI driver for your hardware is not properly installed.

Press OK
Click on the Dial button, enter a phone number and press Place Call
If the connection fails you should check the following:
Is your device properly connected to the telephony network or PBX?
Is the phone number valid?
Do you need a prefix to get an outbound connection?
Can you connect a telephone instead of the hardware, is it working then?
Please refer to your telephony hardware documentation for further troubleshooting.

If the outgoing call succeeds you should now check if you can perform an inbound call
Call the number of the port which is connected to your telephony hardware and see if you get the call signaled and can accept the call
If all tests have succeeded your device should be properly set up for use with TERAVoice

**Question:** I can see my device listed in the Windows Phone Dialer but not in the TERAVoice list of devices, what is the reason?

**Answer:** In order to be suitable for use with TERAVoice it is required that your device can transmit audio. TERAVoice only lists devices that support audio while the Windows Phone Dialer also lists devices that are only used for assisted telephony without audio transmission.

**CAPI Based Hardware**
CAPI based hardware are usually ISDN boards for the European market. TERAVoice supports these boards with its internal TAPI for CAPI driver. This driver only works in combination with TERAVoice.

**Important Note:** The TAPI devices that are installed by the TERAVoice TAPI for CAPI driver will be listed in other applications like the Windows Phone Dialer but they will work only with TERAVoice. Do not use another TAPI application for testing because this will always fail!

In order to test you CAPI based hardware you can use one of the programs that are usually shipped with ISDN boards. If you do not have access to your installation media you can do a search for a CAPI application on the web.

The first attempt and easiest method to be sure that your CAPI based hardware is properly installed is to open the CAPI Configuration (p. 80) dialog and see if your device is listed here.

If your device is listed there and you are still unable to connect the most likely reason is that your connection between the telephony board and the network or PBX is not connected properly.
TERAVoice Server 2004

TERAVoice Does Not Start
There are several reasons why TERAVoice may be unable to start. The first step when trying to identify the reason is always to look into the Event Log (p. 108) and read the most recent error messages.

Some common reasons that cause TERAVoice not to start properly are:

- **3903 - Invalid license information, exiting**
  You need to activate (p. 67) the product first. TERAVoice will not run without activation.

- **3934 - There are more mailboxes defined than the installed license allows. Shutting down**
  Please open the Administration Console (p. 75) and reduce the number of Call Handlers to the amount that your license allows or obtain and install an extension license.

- **3904 - No audio-in devices found! Exiting**
  **3905 - No audio-out devices found! Exiting**
  You do not have a telephony device installed on your system that can be used with TERAVoice.

- **3906 - You do not have sufficient free space on the local hard drive**
  Please free up some disk space and start TERAVoice again.

- **3910 - No Devices were selected**
  Please select at least one device for use with TERAVoice, see Telephony Hardware (p. 77).

The following errors indicate that TERAVoice is not properly installed. Please uninstall (p. 21) and reinstall in order to fix the problem:

- **3900 - Error loading path from registry, exiting**
- **3901 - Error opening configuration information, exiting**
- **3902 - Error loading configuration, exiting**
- **3800 - Error: Unable to find TERAVoice Messages folder**
- **3801 - Error: Unable to find TERAVoice TVShare folder**
- **3802 - Error: Unable to find configuration file**
- **3803 - Error: Unable to connect to configuration**
- **3907 - Error initializing audio files**

TERAVoice Does Not Answer Calls
The first step is to check if the call gets signaled to TERAVoice. Open the Administration Console (p. 75) and select the Server Monitor (p. 107) node.

Call into TERAVoice from a telephone and watch if the call gets signaled. If the call gets signaled please continue below.

**Call does not get signaled**
If you do not see the call being signaled you should check the following items:

- Did you call the correct number?
- Did you select the correct device for use with TERAVoice?
- Are there any other applications open that use the TAPI device in owner mode? If you are unsure about this you can try to stop and restart the Telephony Service and start TERAVoice afterwards.

- Is your telephony hardware connected properly?

- Does it work with a different TAPI application? Follow the instructions in topic Check your telephony hardware (p. 148) to be sure that everything is installed and set up correctly.

- Can you see the message 'Successfully registered calls on '<Your device name>' in the event log?

- Are there any errors being displayed in the Event Log (p. 108)?

**Call gets signaled**

If you can see the call but TERAVoice does not answer it you need to check for the following:

- Do you have a Default Call Handler (p. 66) selected? If the parameters of the call do not match one of the assignment entries for any Call Handler, the call is not answered if you do not have a default Call Handler selected.

- Did you set the option to only answer calls on certain numbers (CalledID's)? Check this setting in the Telephony Hardware (p. 77) configuration dialog and make sure 'Answer all calls' is selected.

- Did you select a Time Schedule (p. 104) Call Handler as default or assigned Call Handler but this Call Handler does not have an action configured for the current date, time and weekday?

- Are there any errors being displayed in the Event Log (p. 108)?
Product Support Options

ONLINE SUPPORT

Product Updates
Product updates are going to be available as fixes and Service Packs which are going to be available from the TERAVoice support page.

We are also going to provide additional sample applications and PlugIns for download. The support page can be accessed via this link:

http://www.terasens.de/fwlink.asp?q=35025

Knowledge Base
For self help support TERASENS provides a Knowledge Base (KB) for TERAVoice on the TERASENS website. The Knowledge Base is going to contain:

- Information about bugs and problems
- How to articles
- Articles with technical background information

You can access the Knowledge Base via the following link:

http://www.terasens.de/fwlink.asp?q=35020

Newsgroups (Usenet)
TERASENS offers free support for TERAVoice Server via public newsgroups. You can access these newsgroups with any newsreader (e.g. Outlook Express). The following newsgroups are available:

General Discussion

news://news.terasens.de/terasens.public.en.teravoice.general

IVR Development

news://news.terasens.de/terasens.public.en.teravoice.development

Note: You can access a German version of these newsgroups by replacing the 'en' with 'de'.

CONTACTING TERASENS

Support by email
Email support is available for all registered customers. Please send a short e-mail message to the following address and follow the instructions in the reply message:

   tvsupport@terasens.de

Support by Phone
Telephone support is available by contract. Please use the telephone numbers provided with your support contract for all inquiries.

Contact Information

TERASENS GmbH
Ackermannstraße 3
80797 Munich
Germany
Phone: +49.89.143370-0 (no support inquiries)
Facsimile: +49.89.143370-22
E-Mail General: info@terasens.de
E-Mail Sales: sales@terasens.de
6 GLOSSARY

A

API: Application Programming Interface
ASP: Active Server Pages - used for creating web applications on Microsoft Internet Information Server

C

Call Handler: A component that offers a certain method for treating a call
Call Transfer: Procedure where A calls B. B creates a consultation call to C while holding A. B then connects/transfer A to C.
CalledID: The phone number of the called party
CallerID: The phone number of the calling party
CAPI: CAPI is short for "Common ISDN API" and is used as API for accessing ISDN hardware. CAPI is mainly used in the European region. More information about CAPI can be found on www.capi.org
COM: Component Object Model - programming model defining object oriented interfaces for code/program libraries
ComISDN: ComISDN TAPI Service Provider for CAPI

D

DTMF: Dual Tone Multi Frequency: Special tones used for dialing or transmitting information

E

extension: In a PBX system or in a public ISDN network every telephone port or line can have one or more extension numbers.

I

IDE: Integrated Development Environment
IIS: Internet Information Server
Inband Signaling: Inband Signaling is a method used by PBXs to inform voice mail systems about the initial target of a call that is redirected to the voice mail system. DTMF tones are used to transmit the information.
IVR: Interactive Voice Response
<table>
<thead>
<tr>
<th><strong>M</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>MSN:</strong></td>
<td>Multi Subscriber Number - used in ISDN networks to enable several phone numbers on a single phone line</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>P</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>pager:</strong></td>
<td>Mobile device that allows reception of numeric or alphanumeric messages</td>
</tr>
<tr>
<td><strong>PBX:</strong></td>
<td>Private Branch Exchange - telecommunication system</td>
</tr>
<tr>
<td><strong>PIN:</strong></td>
<td>Personal Identification Number</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>S</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>SMS:</strong></td>
<td>Short Message Server - allows reception of alphanumeric messages on a mobile phone in cellular networks</td>
</tr>
<tr>
<td><strong>SOHO:</strong></td>
<td>Small office or home office</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>T</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>TAP:</strong></td>
<td>Telelocator Allocator Protocol - used to submit short messages to paging providers or cellular networks</td>
</tr>
<tr>
<td><strong>TAPI:</strong></td>
<td>Telephony Application Programming Interface - Windows-integrated API for controlling telephony devices</td>
</tr>
<tr>
<td><strong>TTS:</strong></td>
<td>Text-To-Speech</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>U</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>UCP:</strong></td>
<td>Universal Computer Protocol - used to submit short messages to paging providers or cellular networks</td>
</tr>
</tbody>
</table>
7 INDEX

A

Access type ................................................................. 122
Action Settings .................................................. 104
Actions.......................................................... 29, 31, 38, 39, 42, 43, 67, 98, 99, 104, 105, 107, 135, 150
Actions depending ..................................................... 38
Activate License .................................................. 22, 90
Activation.......................................................... 21, 22, 40, 90, 148, 150
TERAVoice connects ........................................... 148
Activation fails ....................................................... 148
Activation Methods .................................................. 22
Active Calls List .................................................. 107
Active Server Pages ........................................... 12, 115, 117
ActiveX ............................................................. 31, 38, 41, 101, 122
ActiveX Server .................................................. 122
Add................................................................. 78, 95, 98, 104
Add button .......................................................... 90, 95, 96, 98, 104
Adding .................................................................... 86, 90, 95, 96, 122
Assignment Entries........................................... 95
Address .............................................................. 80
Administration .................................................... 14, 15, 21, 58, 59, 63, 71, 75, 91, 128
Administration Console ........................................... 75
Advanced Calling .................................................. 67
A-law ........................................................................ 80
Alerting ..................................................................... 80
Allocator ............................................................. 57, 86
Allow Recording .................................................... 96
Alphanumeric messages .......................................... 83, 86
Always Simulate Hold ............................................. 91
Analog Networks ................................................... 50, 63
Analog/digital.......................................................... 45
Announce ............................................................. 31, 38, 91, 105, 131, 132, 134
Announce Position On Change ................................ 105
Answer ............................................................. 24, 28, 42, 50, 55, 63, 67, 77, 95, 99, 104, 132, 142, 147, 148, 150
MSN try ............................................................... 63
Wait ...................................................................... 67, 99
Answer ................................................................. 38
Answer All Calls .................................................... 77
A-party ..................................................................... 55, 91

API.......................................................... 14, 15, 18, 21, 24, 30, 33, 34, 37, 41, 52, 53, 55, 62, 63, 67, 76, 77, 80, 83, 101, 105, 107, 115, 117, 138, 142, 147, 148, 150
COM ............................................................... 122
scripting ............................................................. 115, 117
Applications ..................................................... 28, 115, 117, 131, 132
Applications ..................................................... 38
Arrays ..................................................................... 131
Assign ............................................................. 15, 30, 34, 37, 38, 40, 45, 46, 51, 52, 63, 78, 79, 99, 150
Call Handler .......................................................... 38
Assign ................................................................. 38
Assignment ..................................................... 15, 24, 27, 45, 46, 47, 48, 52, 53, 63, 67, 71, 95, 150
device groups ....................................................... 46
Matching ............................................................. 45
Assignment Entries .................................................. 45, 95
Assignment Rules ................................................... 63, 71
Assignment Settings ............................................... 78
AssistantID ............................................................ 132
Assisted Telephony .................................................. 33, 62, 148
ATL COM AppWizard ........................................... 122
ATL Project Wizard .............................................. 122
Attach .............................................................. 15, 28, 122
Attach ................................................................. 38
Audio .............................................................. 59, 80
Audio Delay ........................................................... 62, 78
Audio/Video ........................................................... 62
Authentication Modes ............................................. 91
Auto-Login ............................................................ 47, 103
Automatic Alerting Usage ............................ 80
Automatic Attendant ........................................... 29, 50
Autostart Remote Control Mode ................................ 91

B

Basic Concepts .......................................................... 27
Basic Options .......................................................... 71
Basic Transfer .......................................................... 142
Baud Rate .............................................................. 86
B-Channel ............................................................... 80
Binary Compatibility .................................................. 122
Blind Transfer ....................................................... 34, 55, 142
Borland Delphi .......................................................... 115
BRI .............................................................. 20
Bridged Transfer ....................................................... 55, 91, 98
Bridging ............................................................... 67
<table>
<thead>
<tr>
<th>Topic</th>
<th>Page(s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cycle Modes</td>
<td>43, 105</td>
</tr>
<tr>
<td>Custom IVR Solutions</td>
<td>31</td>
</tr>
<tr>
<td>Custom Setup</td>
<td>21</td>
</tr>
<tr>
<td>ConnectedID</td>
<td>138</td>
</tr>
<tr>
<td>Connection Mode</td>
<td>80, 101</td>
</tr>
<tr>
<td>Contact Information</td>
<td>153</td>
</tr>
<tr>
<td>Conversations</td>
<td>31, 34, 40, 55, 105</td>
</tr>
<tr>
<td>Recording</td>
<td>105</td>
</tr>
<tr>
<td>Country Code</td>
<td>86</td>
</tr>
<tr>
<td>Create Conference</td>
<td>142</td>
</tr>
<tr>
<td>Create Request file</td>
<td>22, 90</td>
</tr>
<tr>
<td>CreateCall</td>
<td>130, 132, 142</td>
</tr>
<tr>
<td>CreateCall function</td>
<td>130, 132</td>
</tr>
<tr>
<td>Creating</td>
<td>12, 24, 31, 45, 47, 71, 91, 104, 115, 117, 122, 130</td>
</tr>
<tr>
<td>ActiveX</td>
<td>31</td>
</tr>
<tr>
<td>ASP</td>
<td>12, 115</td>
</tr>
<tr>
<td>Call Handler</td>
<td>45, 71, 117</td>
</tr>
<tr>
<td>COM</td>
<td>115, 122</td>
</tr>
<tr>
<td>COM Component</td>
<td>122</td>
</tr>
<tr>
<td>H.323 Call Handler</td>
<td>104</td>
</tr>
<tr>
<td>IVR</td>
<td>31</td>
</tr>
<tr>
<td>IVR Script</td>
<td>117</td>
</tr>
<tr>
<td>Outbound</td>
<td>130</td>
</tr>
<tr>
<td>Remote Control Call Handler</td>
<td>47</td>
</tr>
<tr>
<td>Script</td>
<td>117</td>
</tr>
<tr>
<td>voicemail</td>
<td>24</td>
</tr>
<tr>
<td>web applications</td>
<td>117</td>
</tr>
<tr>
<td>Common ISDN API</td>
<td>18, 21, 24, 33, 34, 55, 62, 80, 105, 147, 148</td>
</tr>
<tr>
<td>Common Options</td>
<td>105</td>
</tr>
<tr>
<td>Common Problems</td>
<td>147, 148</td>
</tr>
<tr>
<td>Common Settings</td>
<td>86</td>
</tr>
<tr>
<td>Common Tasks</td>
<td>62</td>
</tr>
<tr>
<td>Component Object Model</td>
<td>31, 101, 115, 122, 132, 135</td>
</tr>
<tr>
<td>Components</td>
<td>24, 30, 31, 38, 41, 83, 101, 107, 115, 122, 132, 135</td>
</tr>
<tr>
<td>Debugging</td>
<td>122</td>
</tr>
<tr>
<td>Device Groups</td>
<td>77</td>
</tr>
<tr>
<td>Device Groups button</td>
<td>46</td>
</tr>
<tr>
<td>Device Groups</td>
<td>37, 46, 67, 77, 142</td>
</tr>
<tr>
<td>Device Features</td>
<td>34</td>
</tr>
<tr>
<td>Device Configuration</td>
<td>77</td>
</tr>
<tr>
<td>Device Settings</td>
<td>79</td>
</tr>
<tr>
<td>Debug Scripts</td>
<td>117</td>
</tr>
<tr>
<td>Default Call Handler</td>
<td>45, 63, 71, 95, 117, 150</td>
</tr>
<tr>
<td>Default Good-Bye Message</td>
<td>58</td>
</tr>
<tr>
<td>Default Greeting Message</td>
<td>58, 63, 76</td>
</tr>
<tr>
<td>Default ISDN API</td>
<td>18, 21, 24, 33, 34, 55, 62, 80, 105, 147, 148</td>
</tr>
<tr>
<td>Direct Call Handler</td>
<td>45, 63, 71, 95, 117, 150</td>
</tr>
<tr>
<td>Direct Call Handler button</td>
<td>46</td>
</tr>
<tr>
<td>Direct Connection</td>
<td>48</td>
</tr>
<tr>
<td>Direct External</td>
<td>48, 71, 79</td>
</tr>
<tr>
<td>Direct Internal</td>
<td>48, 71, 79</td>
</tr>
</tbody>
</table>
TERAVoice Server 2004

I
IDE .......................................................... 115
IDispatch .................................................. 115
Implement Interface.................................. 122
Implementing........................................... 30, 117
Error Handling...................................... 117
IP .......................................................... 30
Implicit .................................................... 80
In-Band Information/Patterns Available....... 80
Inband Signaling............... 15, 48, 51, 63, 71, 79
Activating.............................................. 71
configuring........................................... 63, 79
PBX...................................................... 48
Inband Signaling Options........ 76
Changes ............................................... 76
Indication............................................. 28, 54, 63
Indicator............................................... 12, 15
Initializing............................................ 55
Call Transfer......................................... 55
Insert Default Template..................... 117, 128
Insert menu......................................... 117
Install TERAvoice ...................... 11, 21, 59, 110, 131
Install TERAvoice Client........ 110
Installation.......................................... 21, 110
InstanceCount ...................................... 135
Instantiation........................................ 101
Instantiation Settings....................... 101
Interactive Voice Response.. 12, 14, 15, 31, 34, 38, 39, 41, 61, 91, 101, 115, 117, 130, 135, 138
Interface ........................................... 58, 80, 101, 122
Interface Details................................. 122
Interface setting......................... 80
Interface Type ..................................... 80
Interfaces B-Channel......................... 80
Internal Log ........................................ 15, 91
IP 18, 28, 30, 33, 40, 63, 67, 77, 80, 91, 99
IP address........................................... 63, 67
IP Bridging........................................... 33, 67
ISDN. 12, 15, 18, 20, 21, 24, 27, 33, 34, 45, 46, 47, 52, 55, 57, 62, 63, 77, 78, 80, 86, 105, 107, 147, 148
ISDN board........................................... 105, 148
ISDN Device.......................................... 45, 47
ISDN hardware...... 12, 18, 21, 24, 34, 55, 62, 80, 105, 147, 148
ISDN Interface..................................... 24, 62, 77, 80
IsError................................................. 142
IsOnHold............................................. 138
IsPlayed.............................................. 117
IsPlaying............................................... 138
IsRecording......................................... 138
ITVCallHandler.............................. 122, 135
ITVCallHandler_Init ....................... 122
IVR...... 12, 14, 15, 31, 34, 38, 39, 41, 61, 91, 101, 115, 117, 130, 135, 138, 152
creating............................................ 31
developing........................................... 41, 115
programming....................................... 31
IVR API Reference............................... 135
IVR Applications................................. 115
Developing........................................... 115
IVR Call Handler......................... 101, 135, 138
IVR Handler Log Event...................... 101
IVR Module............................................... 41
IVR Module Call Handler.................... 101
IVR Modules............................ 38, 41, 101, 117, 130, 135
IVR Script............................................. 91, 101, 117
create................................................. 117
debugging............................................. 91
IVR-Module........................................... 101

J
JavaScript............................................. 101, 115, 117
JScript.............................................. 31, 101, 115, 117, 128, 131

K
KB............................................................. 152
Knowledge Base................................. 152

L
LAN.......................................................... 40
Language Support......................... 15, 28, 58
Languages........ 15, 28, 31, 41, 58, 76, 96, 101, 103, 115, 117, 122, 131
Scripting............................................. 117
Languages............................................. 38
Library................................................... 122, 135
Licensing .............................................. 22, 76, 90
Line... 12, 15, 18, 29, 34, 37, 41, 45, 46, 47, 51, 52, 55, 59, 62, 63, 73, 80, 83, 86, 90, 101, 105, 107, 117, 122, 128, 131, 135, 142, 148
MSN..................................................... 63
LineCallOrigin parameter..................... 78
LocalSystem......................................... 59, 122
Logging.............................................. 21, 91, 101
Windows Event Log......................... 91
Logon................................................... 63, 91, 111, 122
LogString............................................. 142

M
Mail Server............................................ 76, 91
Mail Server Settings......................... 76
Mailbox ... 12, 15, 24, 28, 34, 38, 42, 45, 46, 47, 48, 51, 57, 58, 59, 63, 67, 70, 71, 83, 96, 103, 107, 110, 111, 112, 135, 150
Configuring....................................... 71
list......................................................... 42
INDEX

Mailbox Call Handler defines ........................................ 38
Remote control access ................................................ 28
Mailbox ................................................................. 38
Mailbox access ........................................................ 59
Mailbox Call Handler ............................................... 42, 59, 63, 96
Mailbox Call Handler ............................................... 38
Mailbox Configuration .............................................. 12, 70, 71
Mailbox Options ....................................................... 70
Mailbox Folders ........................................................ 59
MSN ................................................................. 15, 34, 45, 46, 47, 52, 63, 80
Multi Subscriber Number. 15, 34, 45, 46, 47, 52, 63, 80
Multi-BRI .............................................................. 20

Multi-PRI ISDN boards .............................................. 12
Music On Hold ....................................................... 41, 43, 55, 101, 105
Music On Hold Call Handler ...................................... 41, 101
MWI ................................................................. 12, 15, 28, 54, 63, 83, 96
MWI Notification .................................................... 15, 28, 83, 96
MWI-Indicator ........................................................ 15

N
Native Call Transfer .................................................. 55
Netmeeting .......................................................... 30, 67
Network Audio Protocol ............................................ 80
Network Configurations ............................................ 45, 50, 63
Networks ......................................................... 50, 52, 57, 80, 83, 86
Newsgroups ......................................................... 152
Newsreader .......................................................... 152
Notification ......................................................... 12, 15, 24, 28, 57, 62, 63, 76, 83, 86, 96, 107, 110
Notification Options ................................................ 76
Notify Queue ....................................................... 107
NT ................................................................. 18
NTFS .............................................................. 59

O
Offline-Activation ................................................... 22
OLE timeouts ........................................................ 122
Online Activation .................................................... 22, 90, 148
Online Help .......................................................... 108
Online Information ............................................... 108, 147
Online Support ..................................................... 152
Outbound ............................................................ 130
Creating .............................................................. 130

P
Pager ................................................................. 28, 57, 83
Pager Notifications ................................................ 28, 57
Paging providers ..................................................... 57
Paging Services .................................................... 57, 86
Pattern Matching .................................................... 71
PBX Inband Signaling .............................................. 48, 71
PBX Outbound ..................................................... 48, 62
PBX ................................................................. 38
PBX Notification ................................................... 63, 83
PCM ............................................................... 73
PDF ................................................................. 110
Perl ................................................................. 31, 101, 115, 117
<table>
<thead>
<tr>
<th>Term</th>
<th>Page Indices</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sending</td>
<td>57, 86</td>
</tr>
<tr>
<td>Sender Number</td>
<td>15, 83, 96</td>
</tr>
<tr>
<td>Supported Audio Formats</td>
<td>59</td>
</tr>
<tr>
<td>Subscriber</td>
<td>34</td>
</tr>
<tr>
<td>Strings</td>
<td>79</td>
</tr>
<tr>
<td>Subscriber</td>
<td>34</td>
</tr>
<tr>
<td>Supported Audio Formats</td>
<td>59</td>
</tr>
<tr>
<td>SMTP Server</td>
<td>63, 91</td>
</tr>
<tr>
<td>SMTP</td>
<td>63, 91</td>
</tr>
<tr>
<td>SMS</td>
<td>57, 86</td>
</tr>
<tr>
<td>SMS/pager</td>
<td>57, 86</td>
</tr>
<tr>
<td>StopPlayRecord</td>
<td>142</td>
</tr>
<tr>
<td>Strings</td>
<td>79</td>
</tr>
<tr>
<td>Subscriber</td>
<td>34</td>
</tr>
<tr>
<td>Stop Play Record</td>
<td>142</td>
</tr>
<tr>
<td>Stop Server</td>
<td>107</td>
</tr>
<tr>
<td>Stop-Bits</td>
<td>86</td>
</tr>
<tr>
<td>Software bridging</td>
<td>142</td>
</tr>
<tr>
<td>Speech</td>
<td>54, 91, 131</td>
</tr>
<tr>
<td>Start Recording Sound</td>
<td>91</td>
</tr>
<tr>
<td>Stop Server</td>
<td>107</td>
</tr>
<tr>
<td>Stop Play Record</td>
<td>142</td>
</tr>
<tr>
<td>Strings</td>
<td>79</td>
</tr>
<tr>
<td>SMS/pager</td>
<td>57, 86</td>
</tr>
<tr>
<td>DTMF</td>
<td>48</td>
</tr>
<tr>
<td>Server Configuration</td>
<td>28, 76, 96</td>
</tr>
<tr>
<td>Server Monitor</td>
<td>107</td>
</tr>
<tr>
<td>Server Options</td>
<td>24, 91</td>
</tr>
<tr>
<td>Service Packs</td>
<td>152</td>
</tr>
<tr>
<td>Service Provider</td>
<td>34, 67, 80</td>
</tr>
<tr>
<td>CAPI</td>
<td>34, 80</td>
</tr>
<tr>
<td>Session</td>
<td>86</td>
</tr>
<tr>
<td>Set up Telephony Devices</td>
<td>62</td>
</tr>
<tr>
<td>Setting MWI Indication</td>
<td>28</td>
</tr>
<tr>
<td>Settings</td>
<td>62, 63, 67, 76, 78, 86, 91, 101, 104</td>
</tr>
<tr>
<td>Set TTS Voice</td>
<td>142</td>
</tr>
<tr>
<td>Setup</td>
<td>11, 14, 21, 28, 41, 45, 50, 58, 59, 110</td>
</tr>
<tr>
<td>Share</td>
<td>110</td>
</tr>
<tr>
<td>TVShare</td>
<td>110</td>
</tr>
<tr>
<td>Simulated Hold Audio</td>
<td>55, 91</td>
</tr>
<tr>
<td>Simulated Transfer</td>
<td>34, 55, 91, 142</td>
</tr>
<tr>
<td>SMS</td>
<td>12, 15, 28, 57, 63, 83, 86, 107</td>
</tr>
<tr>
<td>number</td>
<td>15</td>
</tr>
<tr>
<td>Sending</td>
<td>57, 86</td>
</tr>
<tr>
<td>user's</td>
<td>28</td>
</tr>
<tr>
<td>SMS</td>
<td>38</td>
</tr>
<tr>
<td>SMS And Pager Notification</td>
<td>63, 83</td>
</tr>
<tr>
<td>SMS Log</td>
<td>107</td>
</tr>
<tr>
<td>SMS sending component</td>
<td>107</td>
</tr>
<tr>
<td>SMS/pager</td>
<td>57, 63, 70, 83, 86, 96, 107</td>
</tr>
<tr>
<td>configuring</td>
<td>63</td>
</tr>
<tr>
<td>Diagnostic Tab</td>
<td>57</td>
</tr>
<tr>
<td>queue</td>
<td>107</td>
</tr>
<tr>
<td>sending</td>
<td>57</td>
</tr>
<tr>
<td>SMTP Server</td>
<td>63, 91</td>
</tr>
<tr>
<td>SMTP Server</td>
<td>63, 91</td>
</tr>
<tr>
<td>Snap In</td>
<td>15</td>
</tr>
<tr>
<td>Software bridging</td>
<td>142</td>
</tr>
<tr>
<td>Speech</td>
<td>54, 91, 131</td>
</tr>
<tr>
<td>Start Recording Sound</td>
<td>91</td>
</tr>
<tr>
<td>Stop Server</td>
<td>107</td>
</tr>
<tr>
<td>Stop Play Record</td>
<td>142</td>
</tr>
<tr>
<td>Strings</td>
<td>79</td>
</tr>
<tr>
<td>Subscriber</td>
<td>34</td>
</tr>
<tr>
<td>Supported Audio Formats</td>
<td>59</td>
</tr>
</tbody>
</table>

**Supported Operating Systems**                           | 18
**Supported scripting**                                    | 131
**Syntax-Highlighting**                                     | 128
**SysAudioFolder**                                           | 117, 138
**System Activity**                                          | 107
**Monitoring**                                                | 107
**System Configuration**                                     | 28, 76, 96
**System Manager**                                           | 21
**install**                                                   | 21
**System Requirements**                                      | 18

**T**

<table>
<thead>
<tr>
<th>Term</th>
<th>Page Indices</th>
</tr>
</thead>
<tbody>
<tr>
<td>T112, 37</td>
<td>76, 95, 104</td>
</tr>
<tr>
<td>Take call</td>
<td>76, 95, 104</td>
</tr>
<tr>
<td>Time</td>
<td>76, 95, 104</td>
</tr>
<tr>
<td>TAP</td>
<td>57, 86</td>
</tr>
<tr>
<td>TAPI</td>
<td>12, 12, 18, 20, 30, 33, 34, 37, 52, 53, 55, 62, 63, 67, 76, 77, 80, 83, 105, 138, 147, 148, 150</td>
</tr>
<tr>
<td>TAPI Blind Transfer</td>
<td>55, 142</td>
</tr>
<tr>
<td>TAPI Devices</td>
<td>33, 37, 77, 80, 83, 101, 107</td>
</tr>
<tr>
<td>TAPI Model</td>
<td>80</td>
</tr>
<tr>
<td>Telecommunication</td>
<td>38</td>
</tr>
<tr>
<td>Telelocator</td>
<td>57, 86</td>
</tr>
<tr>
<td>Telelocator Allocator Protocol</td>
<td>57, 86</td>
</tr>
<tr>
<td>Telephone</td>
<td>38</td>
</tr>
<tr>
<td>Telephony Applications</td>
<td>12, 15, 28, 30, 34, 37, 40, 41, 52, 53, 55, 62, 63, 67, 76, 77, 80, 83, 101, 105, 107, 138, 142, 147, 148, 150</td>
</tr>
<tr>
<td>Types</td>
<td>28</td>
</tr>
<tr>
<td>Telephony Device Configuration</td>
<td>24</td>
</tr>
<tr>
<td>Telephony Devices</td>
<td>15, 18, 21, 24, 33, 46, 47, 50, 62, 71, 73, 77</td>
</tr>
<tr>
<td>Telephony Service</td>
<td>150</td>
</tr>
<tr>
<td>TERAVoiceLib.dll</td>
<td>122</td>
</tr>
<tr>
<td>TERAVoiceLib.I TVCallHandler</td>
<td>122</td>
</tr>
<tr>
<td>Test Your Hardware</td>
<td>62, 148</td>
</tr>
<tr>
<td>Testing</td>
<td>62, 117</td>
</tr>
<tr>
<td>Testing Script</td>
<td>117</td>
</tr>
<tr>
<td>TAPI</td>
<td>62</td>
</tr>
<tr>
<td>Testing the Script</td>
<td>117</td>
</tr>
<tr>
<td>Test-Tool</td>
<td>86</td>
</tr>
<tr>
<td>Text-To-Speech</td>
<td>54, 63, 73, 76, 91, 105, 115, 131, 138, 142</td>
</tr>
<tr>
<td>Time Schedule</td>
<td>42, 104, 135</td>
</tr>
<tr>
<td>Time Schedule Call Handler</td>
<td>42, 104</td>
</tr>
</tbody>
</table>

**INDEX**

163
<table>
<thead>
<tr>
<th>Topic</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>Time To Take Call</td>
<td>76, 95, 104</td>
</tr>
<tr>
<td>Timeout</td>
<td>39, 86, 91, 98, 117, 142</td>
</tr>
<tr>
<td>Tone Monitoring</td>
<td>80</td>
</tr>
<tr>
<td>Tones</td>
<td>79, 80</td>
</tr>
<tr>
<td>Tools</td>
<td>111</td>
</tr>
<tr>
<td>Transfer Call</td>
<td>39, 42, 43, 98, 99, 104, 105, 132</td>
</tr>
<tr>
<td>Transfer fails</td>
<td>132</td>
</tr>
<tr>
<td>Transfer Mode</td>
<td>31, 55, 91, 105, 142</td>
</tr>
<tr>
<td>TransferMod</td>
<td>142</td>
</tr>
<tr>
<td>TransferTimeout</td>
<td>142</td>
</tr>
<tr>
<td>Troubleshooting</td>
<td>117</td>
</tr>
<tr>
<td>TsSMSQueue.exe</td>
<td>107</td>
</tr>
<tr>
<td>TTS</td>
<td>54, 73, 91, 105, 115, 131, 138, 142</td>
</tr>
<tr>
<td>Call-Info</td>
<td>91</td>
</tr>
<tr>
<td>include</td>
<td>54</td>
</tr>
<tr>
<td>name</td>
<td>142</td>
</tr>
<tr>
<td>synthesizing</td>
<td>142</td>
</tr>
<tr>
<td>TTS Engine</td>
<td>54, 131</td>
</tr>
<tr>
<td>select</td>
<td>54</td>
</tr>
<tr>
<td>TTS Voice</td>
<td>54, 91, 142</td>
</tr>
<tr>
<td>TvC6Sample1.Calculator</td>
<td>134</td>
</tr>
<tr>
<td>TvC7Sample1.Calculator</td>
<td>134</td>
</tr>
<tr>
<td>Tvsigned_req</td>
<td>22, 90</td>
</tr>
<tr>
<td>TvC1Sample1.Calculator</td>
<td>132</td>
</tr>
<tr>
<td>TvC1Sample2Features</td>
<td>132</td>
</tr>
<tr>
<td>TvC6Sample3.CallDistribution</td>
<td>132</td>
</tr>
<tr>
<td>TvC6Sample4.Wakeup</td>
<td>132</td>
</tr>
<tr>
<td>TVShare</td>
<td>59, 110</td>
</tr>
<tr>
<td>Type Library</td>
<td>122, 135</td>
</tr>
<tr>
<td>TypeLib Description</td>
<td>122</td>
</tr>
<tr>
<td>TypeLib ID</td>
<td>122</td>
</tr>
<tr>
<td>UCP</td>
<td>57, 86</td>
</tr>
<tr>
<td>UCP Operation</td>
<td>86</td>
</tr>
<tr>
<td>Unanswered Calls</td>
<td>42</td>
</tr>
<tr>
<td>UnHoldCall</td>
<td>142</td>
</tr>
<tr>
<td>Unified Messaging</td>
<td>12, 28</td>
</tr>
<tr>
<td>Unimodem/5</td>
<td>33, 34, 62</td>
</tr>
<tr>
<td>Uninstalling TERAVoice</td>
<td>21</td>
</tr>
<tr>
<td>Universal Computer Protocol</td>
<td>57, 86</td>
</tr>
<tr>
<td>Unreachable Action</td>
<td>99</td>
</tr>
<tr>
<td>Updates</td>
<td>152</td>
</tr>
<tr>
<td>Using</td>
<td>75</td>
</tr>
<tr>
<td>Administration Console</td>
<td>75</td>
</tr>
<tr>
<td>CAPI Devices</td>
<td>24</td>
</tr>
<tr>
<td>Redirector</td>
<td>47</td>
</tr>
<tr>
<td>Remote Control</td>
<td>70, 91</td>
</tr>
<tr>
<td>Remote Control</td>
<td>38</td>
</tr>
<tr>
<td>TERAVoice</td>
<td>11, 14, 61, 110, 148</td>
</tr>
<tr>
<td>V</td>
<td></td>
</tr>
<tr>
<td>VB</td>
<td>122</td>
</tr>
<tr>
<td>VBScript</td>
<td>31, 101, 115, 117, 128, 131</td>
</tr>
<tr>
<td>Version Information</td>
<td>76, 94</td>
</tr>
<tr>
<td>Visual Basic</td>
<td>122, 132, 134</td>
</tr>
<tr>
<td>samples</td>
<td>132</td>
</tr>
<tr>
<td>Visual Basic Debugger</td>
<td>122</td>
</tr>
<tr>
<td>Visual Basic Samples</td>
<td>132</td>
</tr>
<tr>
<td>Visual C++</td>
<td>122, 134</td>
</tr>
<tr>
<td>Voice boards</td>
<td>33</td>
</tr>
<tr>
<td>Voice Mail</td>
<td>48, 51, 54, 63, 71, 76, 79</td>
</tr>
<tr>
<td>Voice Mailbox</td>
<td>24, 28, 38, 39, 42, 43, 45, 50, 63</td>
</tr>
<tr>
<td>Voice Menu</td>
<td>29, 31, 39, 42, 76, 91, 98, 135</td>
</tr>
<tr>
<td>Voice Menu Language</td>
<td>76, 91</td>
</tr>
<tr>
<td>Voice Message</td>
<td>28, 63, 96, 98, 101</td>
</tr>
<tr>
<td>Voice Message</td>
<td>38</td>
</tr>
<tr>
<td>Voice Modems</td>
<td>20</td>
</tr>
<tr>
<td>Voice Over IP Gateway</td>
<td>30</td>
</tr>
<tr>
<td>Voice Prompt Language</td>
<td>58, 63</td>
</tr>
<tr>
<td>Voice prompt telling</td>
<td>70</td>
</tr>
<tr>
<td>Voice Prompts</td>
<td>29, 58, 63, 70, 73, 96, 103</td>
</tr>
<tr>
<td>Configuring</td>
<td>73</td>
</tr>
<tr>
<td>Voicemail</td>
<td>12, 24, 28, 34, 47, 48, 51, 63, 71, 79, 83</td>
</tr>
<tr>
<td>creating</td>
<td>24</td>
</tr>
<tr>
<td>delay</td>
<td>71</td>
</tr>
<tr>
<td>PBX</td>
<td>48</td>
</tr>
<tr>
<td>Voicemail</td>
<td>38</td>
</tr>
<tr>
<td>Voicemail application</td>
<td>71, 83</td>
</tr>
<tr>
<td>running</td>
<td>83</td>
</tr>
<tr>
<td>Voicemail Settings</td>
<td>63</td>
</tr>
<tr>
<td>VoiceName</td>
<td>142</td>
</tr>
<tr>
<td>VoIP</td>
<td>20, 30, 38, 40, 80, 98, 104</td>
</tr>
<tr>
<td>use</td>
<td>38</td>
</tr>
<tr>
<td>W</td>
<td></td>
</tr>
<tr>
<td>Wait Before Taking Call</td>
<td>79</td>
</tr>
<tr>
<td>Wait Between Tones</td>
<td>79</td>
</tr>
<tr>
<td>Wait for answer</td>
<td>67, 99</td>
</tr>
<tr>
<td>Wait for components</td>
<td>122</td>
</tr>
<tr>
<td>Wait Until Answer</td>
<td>91</td>
</tr>
<tr>
<td>Wait Until Connect</td>
<td>55, 105</td>
</tr>
<tr>
<td>Waiting Queue</td>
<td>34, 43, 46, 105, 135</td>
</tr>
<tr>
<td>Waiting Queue Call Handler</td>
<td>34, 105</td>
</tr>
<tr>
<td>Waiting Queue Viewer</td>
<td>105</td>
</tr>
<tr>
<td>WakeUp</td>
<td>132</td>
</tr>
<tr>
<td>WakeUp System</td>
<td>132</td>
</tr>
<tr>
<td>WakeupCall</td>
<td>130</td>
</tr>
<tr>
<td>WakeupCall sample application</td>
<td>130</td>
</tr>
<tr>
<td>Term</td>
<td>Page Numbers</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>--------------</td>
</tr>
<tr>
<td>WAV-format</td>
<td>59</td>
</tr>
<tr>
<td>Weekdays</td>
<td>28, 29, 63, 96, 104, 150</td>
</tr>
<tr>
<td>Windows Event Log</td>
<td>91</td>
</tr>
<tr>
<td>Windows Installer</td>
<td>110</td>
</tr>
<tr>
<td>Windows Phone Dialer</td>
<td>30, 62, 67, 148</td>
</tr>
<tr>
<td>Windows Scripting Host</td>
<td>31, 117</td>
</tr>
<tr>
<td>WriteCallLog</td>
<td>117, 122</td>
</tr>
<tr>
<td>WriteLogFile</td>
<td>142</td>
</tr>
</tbody>
</table>