Vox Studio
User Manual

Xentec

Namenstraat 11, Nossegem
B-1930 Zaventem
Belgium
Tel +32-2-757-0666
Fax +32-2-757-0666
Email 100276.344@compuserve.com

Table of Contents

Introduction to Vox Studio

Vox Studio is a professional software tool from Xentec nv-sa. It facilitates the efficient production of numerous digitized message files for voice processing applications like Voice Mail, Interactive Voice Response, Phone Banking, Audiotex, and others. Most of the tedious tasks related to the production of prompt files are now automated and produce consistent high-quality results.

Thanks to Vox Studio these message files (prompts) can now be prepared in-house using a PC, a multimedia-compliant sound card (for example SoundBlaster or compatible) and a decent quality microphone. Vox Studio records into the usual multimedia (.wav) file formats.

Vox Studio can convert Windows .wav files into telephony files like ADPCM, A-law PCM, Mu-law PCM, linear PCM and vice versa. You can convert from any of the above formats to any other of the above formats.

Prompt recording can be assisted by a tele-prompter function built into Vox Studio. The tele-prompter flashes the text of messages one by one on the PC's screen so that it becomes very easy and time-efficient to record a large number of prompts for a new application or add new messages to an existing application.

Another time-consuming task automated by Vox Studio is the clean-up of recorded prompts through addition and removal of silence at the beginning and end of the files. Vox Studio can do this automatically for you and standardize to the length of silence that suits your application best.

There are a lot of goodies built right into Vox Studio, like reduction of talk-off risk through DTMF attenuation or automatic volume level normalization of all files for maximum intelligibility, and more. Voice Processing professionals will immediately recognize the advantage of having all these capabilities regrouped into one, easy-to-use application.

Vox Studio is a user-friendly Windows 3.1 application and requires a sound card only if files are to be recorded or played back. No sound card is necessary if Vox Studio is exclusively used to convert from one voice file format to another or do filtering operations.

The price has been kept low. Vox Studio can therefore be used by application developers and by their customers, such as voice system administrators or end users wishing to do regular voice file maintenance.

The best way to find out more about Vox Studio is to read the manual, browse through the help file (voxtudio.hlp) or to switch on your Multimedia PC and start playing with Vox Studio itself.

Vox Studio Functionality

Vox Studio is a powerful tool for anyone needing regular production of top quality voice prompts to be used in voice processing and telephony applications.

Vox Studio provides Graphical (cassette recorder-like) tools to record and play back prompt files in various formats at several standard sampling rates. You need a Windows 3.1 PC equipped with Vox Studio and a multimedia compliant sound card to record or play back. You do not need a sound card for conversions only.

Conversions are possible between all multimedia and voice processing telephony formats known by Vox Studio. Conversion from stand-alone files to concatenated indexed files and back is possible too.

The tele-prompter is similar to the tool that TV newspeople and public speakers use to read their announcements from.

Vox Studio can generate files containing pure DTMF tones of various length.

Special effects such as high-pass and low-pass filtering, DTMF filtering, signal centering around the base-line, silence and volume normalization can be applied.

Most of these functions can be performed in automated "background" mode so that you can continue to use your PC for other tasks (like code editing or even for non-batch functions in

Vox Studio itself). You could, for instance, listen to prompts in Vox Studio while file conversions occur in the background ! Talking about productivity ? To find out the full details of what Vox Studio can do for you browse through the subjects below:

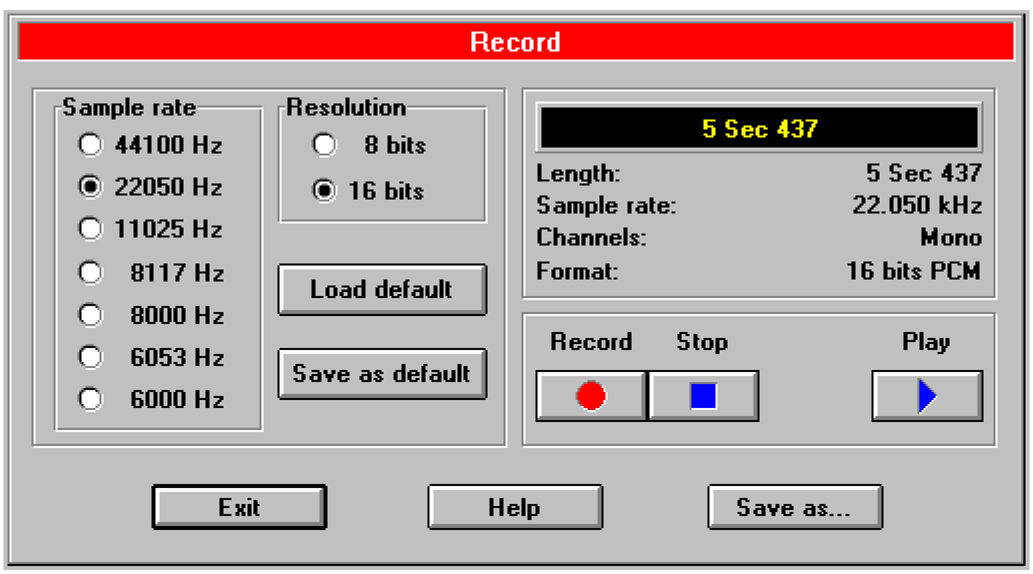
Recording functionality

A graphical user interface allows simple recording of 8-bit and 16-bit multimedia .wav files at frequencies ranging from 6 kHz to 44.1 kHz.
Recording is done by clicking on the buttons of a user-friendly cassette recorder-like device. The prompts are recorded one by one as .wav files and can be played back immediately for verification.

Record command

Opens a cassette recorder-like graphical window. From there you can record exactly as if you were using a real recorder. You can record, stop, pause and playback.
Before starting a recording you should define the resolution (8 or 16 bits) and sampling rate (6 to 44 kHz) of the .wav file you wish to record.
After recording a prompt you can play it back or save it in the previously defined format. Once a prompt is saved, Vox Studio is ready for a new recording and cannot play a file back from the recorder window.
Vox Studio always records .wav files. You can later convert these into any format known to Vox Studio.

Recorder window



On the left side you indicate the sample rate and resolution of the file you are going to record.
Using the cassette recorder-like buttons on the right side you can start and stop recording.
The play button opens the playback window, so that you can listen to your recording.

Playback functionality

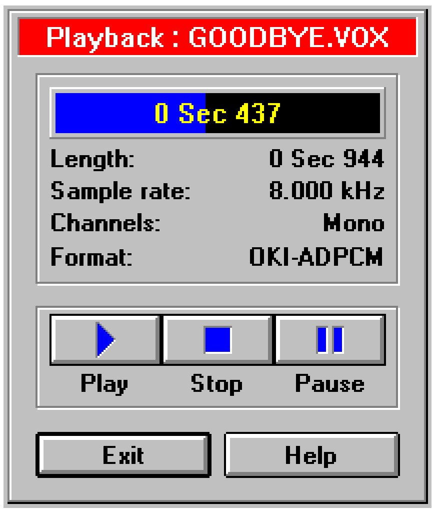
A graphical user interface allows simple playback of mono or stereo 8-bit and 16-bit

multimedia .wav files at sample frequencies ranging from 6 kHz to 44.1 kHz and of telephony .vox files in ADPCM, A-law, Mu-law and linear PCM at sample frequencies ranging from 6 kHz to 8.1 kHz.
Playback is activated by clicking on the buttons of a user-friendly cassette player-like device.

Playback command

Allows playback of a previously opened file, in any format known to Vox Studio, or of the current recorded prompt if activated from within the recorder window.

Playback window



The playback window allows you to play the currently loaded file. This can be any file loaded using the File/Open menu commands or the result from the previous sound file conversion, if that file is still loaded. You can check which file is current by looking at the title bar. If the file name is Untitled, then no file is currently open. Otherwise, the title bar will give you the name of the currently open file. This file can be played by clicking on the Play button.
The display indicates the file duration in seconds and milliseconds, the sampling rate in kHz, the recording format used and the stereo/mono status (for wave files).

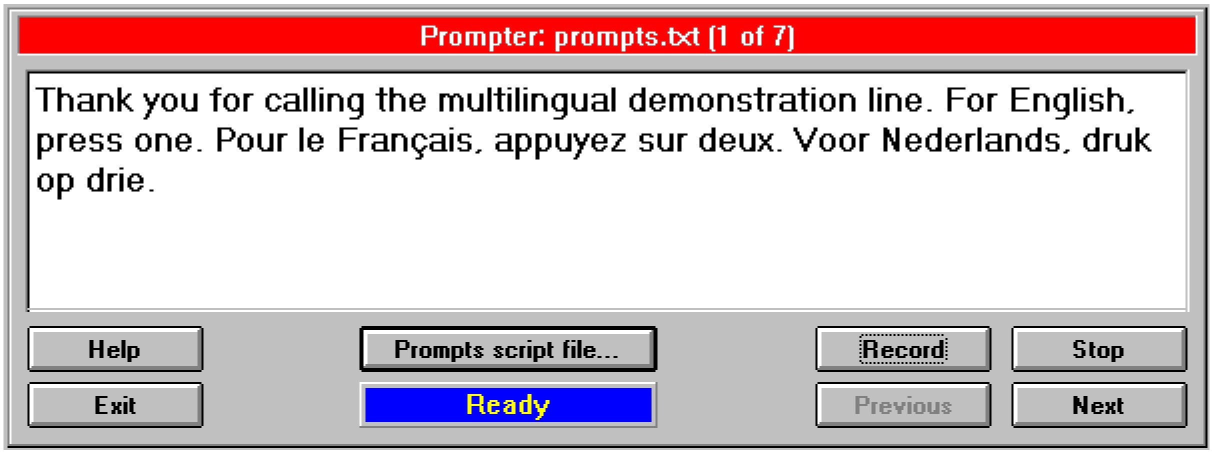
Teleprompter functionality

A graphical teleprompter allows simple recording of 8-bit and 16-bit multimedia .wav files at sampling frequencies ranging from 6 kHz to 44.1 kHz.
Based on your ASCII text file script, prompt messages are flashed one by one the PC's screen and can be recorded by tapping the space bar or clicking a button and simply reading the messages as they appear on the screen.
The prompts are automatically recorded one by one as .wav files with the names pre-defined in your ASCII text script file.
More script formats may be added in the future, to ensure direct compatibility with popular voice processing application generators.

Prompter command

The prompter allows you to load an ASCII text file containing a series of prompts and associated file names.
The prompter flashes the messages on screen one by one and enables the speaker to record them and save them automatically under the file names defined in the ASCII script file.

Prompter Window



The prompter allows you to load a prompt script file which contains information such as the prompt texts, the name of the files to save the prompts in and a short description for each prompt. You can read and record the prompts one by one from the prompter screen simply by tapping the space bar. The space bar initiates and stops each recording. The Left and Right arrow keys allow navigation from prompt to prompt while the Up and Down or Page Up and Page Down keys allow scrolling through a long prompt that does not fit on a single screen. It is best to avoid such long prompts. Prompts can be re-recorded easily.
The format of the prompt script file is the same as the format used by the Group and Ungroup Indexed commands. Therefore, a recording sequence made with the Vox Studio prompter and translated into the final target format can immediately be used to generate one large indexed file containing all the concatenated recorded prompts.

DTMF Generation functionality

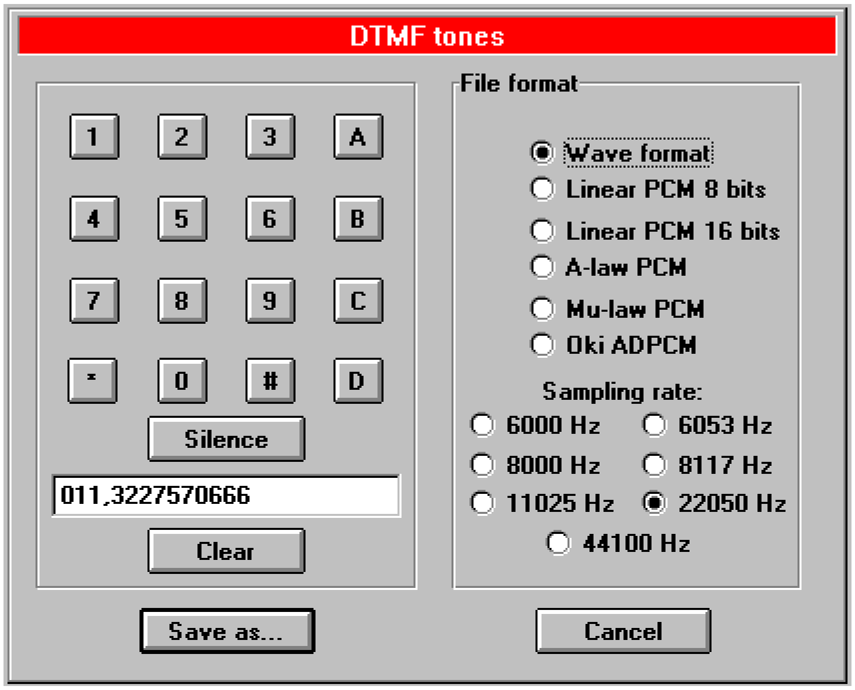
The graphical representation of a portable telephone keypad (beeper) allows keying in a string of DTMF digits and pause commands.
DTMF digits 0 to 9, A to D and * and # are all available.
The default length of the produced DTMF tones and silences is programmable.
DTMF sequences can then be saved in various file formats.

DTMF Generation command

The DTMF file generator is presented under the appearance of a telephone keypad.
You can enter a DTMF sequence using all 16 DTMF tone pairs, including the A, B, C, D codes. A sequence is generated by simply clicking on the corresponding keys on the screen.
One can also add pause commands (represented in the display by a comma) in the DTMF sequence.
The length of all DTMF tones is as programmed by default through the Default Miscellaneous menu.

The DTMF sequences can be saved as sound files. Higher sample rates produce more accurate DTMF tones.

DTMF Generation window



Compose a DTMF dial string on the keypad and save it in the desired file format. Avoid using 6 kHz sample rates for DTMF because of the inherently large frequency inaccuracies at such a low sampling rate.
Enter a sequence of digits by clicking on the appropriate digit keys. The digit sequence, including pauses (represented by commas), appears in the left readout window.
Select the format under which you want to save the generated DTMF sequence.
When generation ends, you will be prompted for the file name and path under which the generated DTMF sequence will be saved.

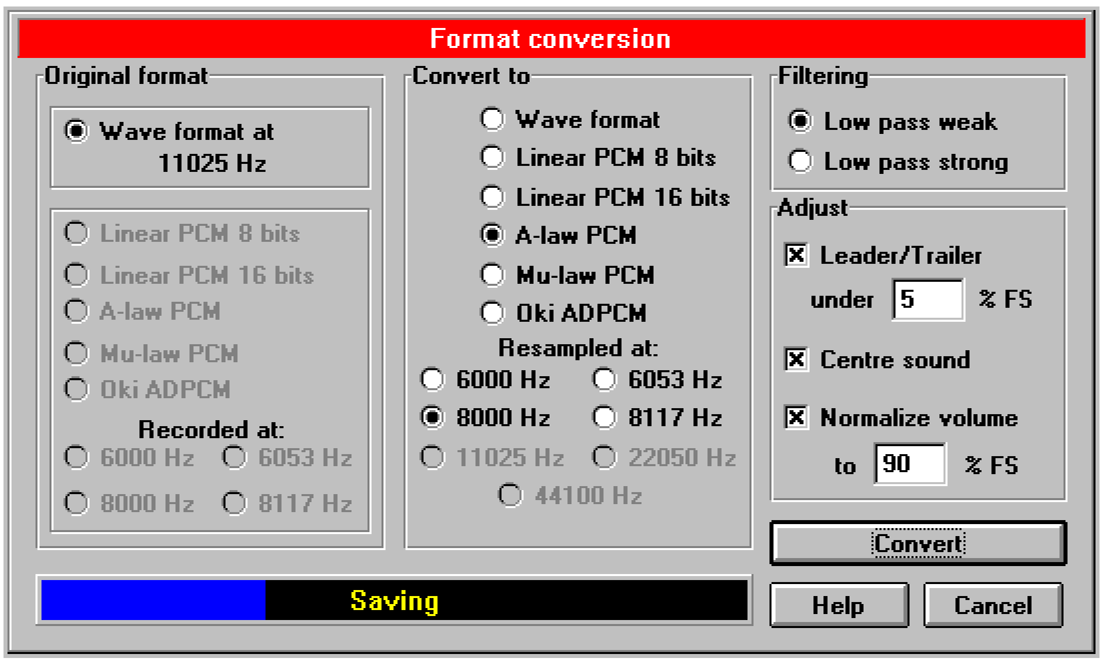
Conversion functionality

Vox Studio can convert any file format it knows to any other file format it knows. This includes the capability to change the sampling rate (down and up) as well as the coding algorithm ! You can, for instance, convert a 44.1 kHz .wav file into a 6.053 kHz ADPCM file or convert a Mu-law PCM file at 8 kHz into an A-law PCM file at 8 kHz. You could even do unusual operations, like converting a 6 kHz ADPCM file into a 22 kHz wav file !
The amplitude of the recorded signals is normally left unchanged by the conversion processes. You can, however, elect to activate the "Normalize Volume" option in the conversion dialog boxes or use the Normalize menu command. This will give you the ability to do a posteriori automatic amplitude adjustments.
See the topic Functionality Matrix for a complete summary of what files Vox Studio can record, playback, convert or otherwise manipulate.

Convert Current command

The Convert Current dialog box allows you to select the convert from and the convert to file format and sample rate. You need to open a file before you can do a manual conversion. You can convert from any format known to Vox Studio to any other format known to Vox Studio. You can do up conversions and down conversions. See File formats used by Vox Studio for an enumeration of formats Vox Studio currently handles.
You always need to fully describe the format you want to convert to. You need to describe the format you want to convert from only if the source file is not a .wav file.
A nice feature of Vox Studio is that you can do file conversions without having a sound card installed in your system. Of course, you need a card to record or play any file.
While converting you can also adjust leading and trailing silence or normalize the sound amplitude. See further in this document for a detailed explanation of what the Leader/Trailer and Normalize commands do.
Select a strong or a normal conversion filter, do this by listening to a typical converted file played on the final target sound card or telephony card.

Convert Current window



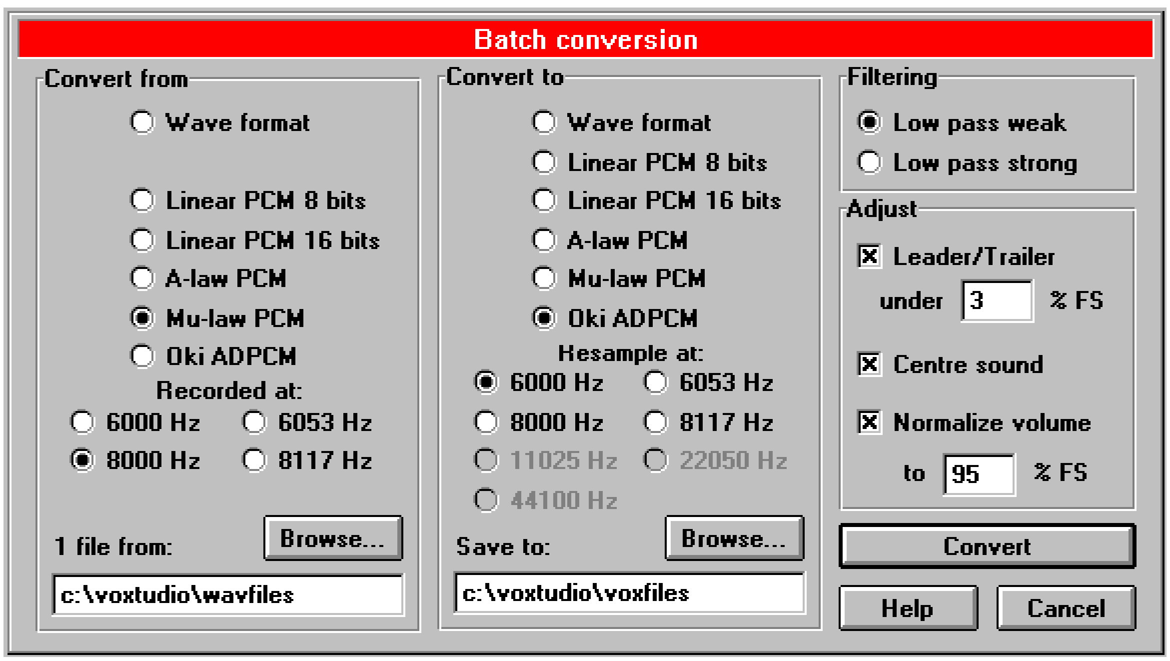
Select both a "convert from" and a "convert to" file format for manual conversion. The exact details of a "convert from" file in wave (.wav) format need not be entered as Vox Studio reads that information from the wave file header. You do, however, need to specify the frequency for a "convert to" wave format. If the convert from format is a telephony format you need to enter all format details.
The conversion source file is the currently open file.
When the conversion ends you will be prompted for the name and directory under which the converted file should be saved.
Leader/Trailer silence adjustment and volume normalization are optional functions which can be performed on-the-fly while converting.

Batch Conversion command

A dialog box allows you to select the convert from and the convert to file format and sample

rate. You can convert from any format known to Vox Studio to any other format known to Vox Studio. You can do up conversions and down conversions. See File formats used by Vox Studio for a complete enumeration of formats Vox Studio currently handles.
You always need to fully describe the format you want to convert to. You need to describe the format you want to convert from only if the source file is not a .wav file. All source files should have the same format. All converted files will have the same format.
Batch conversion allows you to select multiple files (up to 1024 files) or a complete directory to convert. Use the usual Windows file selection techniques to indicate which files need converting.
A nice feature of Vox Studio is that you can do file conversions without having a sound card installed in your system. Of course, you need a card to record or play any file.
While converting you can also adjust leading and trailing silence or normalize the sound amplitude. See further in this document for a detailed explanation of what the Leader/Trailer and Normalize commands do.
Batch conversion will occur in the background so that, if desired, you can do some other work under Windows on your PC (including using Vox Studio for other tasks).
Select a strong or a normal conversion filter, do this by listening to a typical converted file played on the final target sound card or telephony card.

Batch Conversion window



Select both a "convert from" and a "convert to" file format for batch conversions. The exact details of "convert from" files in wave (.wav) format need not be entered as Vox Studio reads that information from wave file headers. You do, however, need to specify the frequency for "convert to" wave formats. If the convert from format is a telephony format you need to give format details.
Specify what the conversion source files are and in which directory the converted files are to be saved.
You can convert selected files or entire directories. Selection is done by browsing through your directories and files. You can select an entire range of files by pressing the SHIFT key while clicking on the first and last file of the range. You can select many individual files by pressing the CTRL key while clicking on files.

Leader/Trailer silence adjustment and volume normalization are optional functions which can be performed on-the-fly while converting.
Batch conversion is a background process. You can continue to use Vox Studio or Windows for other work while conversion is being done in the background.

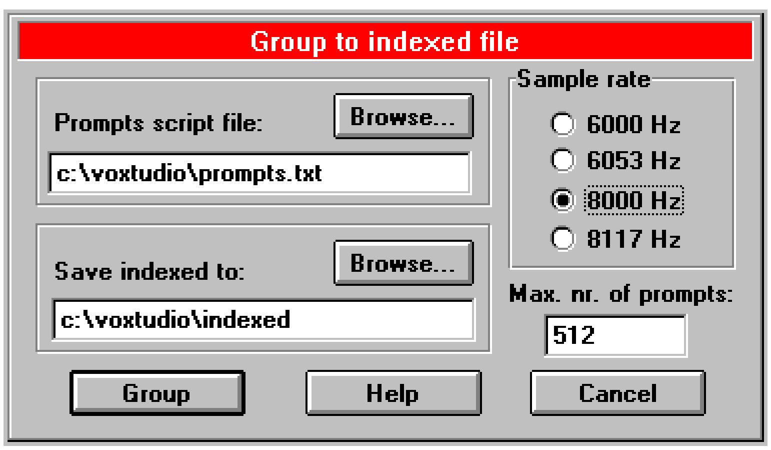
Group and ungroup functionality

Grouping: Individual, stand-alone prompt files can be converted into one, single, large indexed file. This automated procedure is directed through a script file. The tele-prompter script file and the group and ungroup script files are all compatible with one another.
Ungrouping: Indexed files can be expanded into their individual components. An optional script file can be produced. This script file is compatible with the tele-prompter script file and the group script file.
Thanks to the script file it is now possible to smoothly record individual prompts with the tele-prompter, group them into an indexed file, ungroup the files and re-record one of the prompts then re-group it all to an index file, etc.

Group Indexed command

Group Indexed allows concatenation of several stand-alone telephony voice files (often with a .vox extension) into one single indexed telephony voice file (often with a .vap extension)

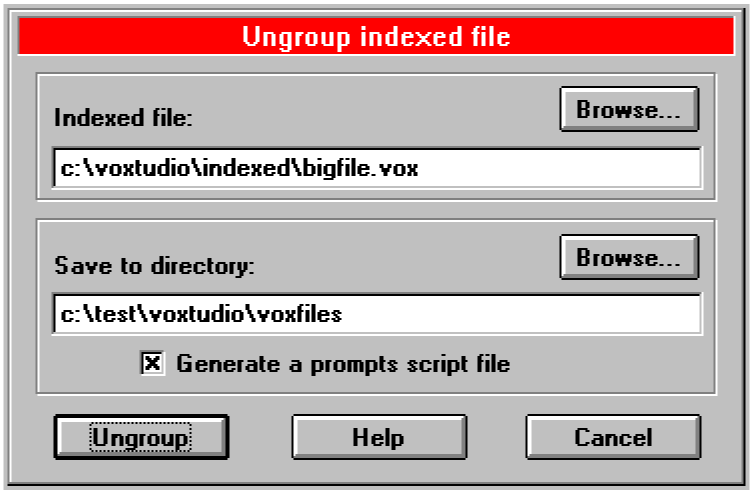
Group Indexed command window



Ungroup Indexed command

Allows expansion of an indexed telephony voice file (usually .vap) into its stand-alone components (usually .vox).

Ungroup Indexed window



Center functionality

Vox Studio has the capability to re-center a sound signal around the zero baseline if, for whatever reason, a DC offset or a very low frequency interference has been added to the original sound signal.
This restores the signal's symmetry, and is useful if the signal has to be amplified or otherwise manipulated.

Center command

This command automatically re-centers the sound signal around the zero base-line to eliminate DC offsets, and very low frequency interference.
When a Normalize command is issued, a Center command is in fact performed automatically on the signal prior to amplitude normalization.

Amplitude normalization functionality

For purposes of intelligibility and consistency, menu prompts and recorded messages should all be played loud and clear and with similar sound volumes to the telephone line.
It is not always easy to assemble several hundred messages for a telephony voice processing application and have them all recorded at nearly constant and nearly maximum amplitude.
This again is where Vox Studio can help. Vox Studio can, as an option, automatically normalize the amplitude of all your files. Sit back and relax while Vox Studio does the dirty work for you.

Normalize command

This superb productivity feature allows easy production of prompt files with equal (and, for telephony cards, preferably high) volume levels.
A dialog box allows you to select the percentage of "full scale" amplitude desired. The current file will be scanned for maximal sound level and the whole file will be multiplied by a factor that will bring the maximum sound volume to the desired % of "full scale".
When a Normalize command is given a Center command is in fact performed automatically on the signal prior to amplitude normalization.

Silence adjustment functionality

Many voice processing professionals buy expensive voice editors that can do nice cut and paste operations on sound files and then end up doing only one thing with their voice editor: manually add or remove periods of silence at the beginning and end of their recordings. This is usually a time consuming and boring trial-and-error activity.
You can automate this repetitive task with the help of Vox Studio. You select a default length for leading and trailing silence and give a single command to apply it to your prompt. Vox Studio goes and does the work for you.
Automatic silence adjustment is a threshold-activated process and requires spotless, clean recordings.

Leader / Trailer command

This superb productivity feature allows easy production of prompt files with equal leading and trailing blanks.
Automatic silence adjustment is a threshold-activated process and requires spotless, clean recordings.
A dialog box allows you to select the percentage of "full scale" amplitude used as a threshold to detect silence. The current file will be scanned for sound level and Vox Studio will automatically detect the beginning and end of sound in the file. It will then adjust the length of leading and trailing silence to a fixed number of milliseconds, which can be the same for all your files.
The silence duration is defined through the Default Miscellaneous menu.

DTMF filtering functionality

Sometimes you can have a recorded file with human voice or music that repeatedly causes voice processing hardware to erroneously recognize DTMF sounds on the telephone line. This is called talk-off. What happens, in fact, is that the voice of the speaker or the music being played contain pairs of frequencies very similar to the DTMF frequencies generated by telephone keypads. This happens only once in a while, and only on a small number of files. Vox Studio can fix this for you by removing the frequencies close to DTMF frequencies from your sound file. One pass through Vox Studio's DTMF attenuator usually fixes this. Of course, removing sound at various places in the frequency spectrum does affect sound quality. So, use this only when you need it, and only on the files that need it. We would advise you to experiment with this function before you start using it on hot data.

Filter command

The filter dialog box allows individual selection or deselection of a high-pass, low-pass and DTMF attenuation filter. In addition, for the high-pass and low-pass filters the -3 dB cut-off frequencies can be chosen.
The Nyquist theorem says: the highest frequency component in any sampled file cannot exceed 0.5x the frequency at which the file is sampled. Aliasing errors occur whenever the Nyquist theorem is overlooked. It makes little sense to select a cut-off frequency above 0.5x the sampling frequency of the file.
The DTMF filter removes signal components at and near the frequencies which correspond to the DTMF tone-pair frequencies. This harsh treatment effectively removes talk-off problems, but also somewhat degrades the sound quality. Use this only on files that really do cause talk-off problems.

High & low pass filtering functionality

The high-pass filter has a programmable cut-off frequency. It rapidly removes frequency components from your sound file below a selected frequency.
The low-pass filter also has a programmable cut-off frequency. It rapidly removes frequency components from your sound file above a selected frequency.
Use these filters only if needed and if you know precisely what you are doing and how this is going to affect your digitized files. Always test filtering and other effects you apply to valuable sound files before you get rid of the originals.

Installation and customization

System requirements

Windows and DOS requirements

Vox Studio is a Windows 3.1 application.
You need to have your Windows sound card drivers installed prior to using the recording and playback capabilities of Vox Studio. Consult the installation manual of your multimedia sound card for driver installation details. Test the installation of your sound card with the software that came with the card before attempting to use Vox Studio.
If you have the possibility to install the Microsoft Sound Mapper as well, it may be a good idea to do so. Sound Mapper allows you to play 16 bit files even if your sound card is only an 8-bit card or stereo files on mono systems. Vox Studio can play and manipulate stereo source files but, of course, always generates mono files for telephony purposes.

CPU and speed requirements

Vox Studio’s algorithms are tuned for accuracy and speed. Vox Studio does a huge number of calculations (tens of millions) for every conversion or filtering. This is done using the PC's CPU and no additional hardware is required by Vox Studio.
A 386 with a 387 co-processor is an acceptable platform for Vox Studio. Of course, the faster your PC, the better. A fast 486 or Pentium is the ideal platform for Vox Studio to really fly. As an example, with a 66mhz 486, filtering is performed almost in real time.

Memory requirements

In order to run on any Windows 3.1 PC, Vox Studio does not rely exclusively on available memory. If it were, it would be impossible to manipulate very large voice files. Vox Studio uses your hard disk and temporary files as a scratch-pad area. This is invisible to the user, except for the obvious disk space requirement.
Still, for Windows 3.1 to run comfortably without too much disk swapping, we recommend a minimum of 8 Mbytes of RAM. You can do with less, you cannot be very happy with less. This is really a Windows performance requirement, not a Vox Studio need.

Display requirements

Vox Studio was written for and tested with VGA and SVGA configurations under Windows 3.1 and 3.11.
We do not know how it behaves with other display configurations. We cannot guarantee full functionality with odd display configurations, but we really do not expect any problems.

Disk space requirements

Disk space is an important requirement for Vox Studio to run satisfactorily. Although Vox Studio will warn you when it cannot continue due to low disk space, it is good practice to always check your available hard disk space before starting a lengthy recording project or conversion session.
Naturally, for conversions, Vox Studio needs enough free disk space to store both the original voice files and the conversion results plus a few temporary scratch-pad files. This does NOT mean twice the size of the original files ! It all depends on the sampling rate and format used. Remember that 16-bit files at 44.1 kHz take up 30 times more disk space than the same prompt in 6 kHz ADPCM format !

Sound card requirements

You need a multimedia-compliant sound card if you want to use Vox Studio to record or play back.
No multimedia sound card is needed if Vox Studio is used to perform format conversions only.
A sound card is not a must for playing back files if you have a set-up to play files over your voice processing telephony card. You will not be able to play back .wav files that way though.
Select a card that can do 16-bit recordings and can handle sample frequencies of 44 kHz down to 6 kHz.
Select a card that comes with a Windows sound mixer and sound editor.

Microphone requirements

A directional microphone is the best companion to Vox Studio. It helps reduce stray noises from the PC's power supply fan, coughing office colleagues, passing aeroplanes, the photocopier, etc. You don't need a very expensive microphone, just a very directional one. The poor transmission quality of the telephone networks does not justify an expensive studio microphone. A directional microphone that picks up nothing else but your spoken words and no background noise is, however, a real blessing.
You may have to experiment. Try and find a reasonably sound-proof area in your office so that outside noises and PC-hum is minimized. All those spurious sounds could otherwise end up recorded in your prompts.

Headphones

Although not really a requirement, as sound files can be played over your usual multimedia speaker(s), headphones are the best tools to evaluate the quality of message files you have just recorded or converted. Watch those background noises and slight breathing sounds.

Set-up

Personalize your Vox Studio

Personalization of your Vox Studio program occurs during the installation process. It is important to provide all the information requested by the setup program. The setup program is the only way to install Vox studio correctly.

Create directories for Vox Studio

You will most probably use a lot of directories in order to keep your sound and program files well organized. We suggest you create at least three directories:
One of them will hold your Vox Studio, and associated, program files.
The second could be the default directory to hold the multimedia .wav files.
The third could be the default directory to hold the telephony .vox files.

Installing and Copying files

To install, start Windows, from Program Manager select File/Run and type A:\setup, then click on OK.
You cannot manually copy the program files, you have to go through the installation process using A:\setup.exe for Vox Studio to work.
The installation process copies all necessary files to your Vox Studio directory.

Add Vox Studio to Program Manager

The installation procedure normally takes care of this. Just in case you do lose your Vox Studio icon by accident, you can re-create a program icon as follows:

1. Go into Program Manager.
2. Open the application group in which the new Vox Studio icon will reside.
3. Click on File/New/Program item/OK.
4. In the program item dialog box select Browse.
5. Navigate to the directory containing the file voxtudio.exe.
6. Select the file and press OK, then OK again. A new Vox Studio icon will be created in the open application group.

Set Vox Studio defaults

You can set all Vox Studio defaults from the Defaults menu. At the very least you should set the default directories, file extensions and sound devices or Vox Studio will not work correctly.

Defaults menu

We strongly suggest you browse through the various default configuration options in the Defaults menu and set the defaults to your liking before you start using Vox Studio.
At the very least you should set the default directories, file extensions and sound devices or Vox Studio will not work correctly.

External Programs command

This command allows selection of the 4 programs that will appear in the Utility menu.
You can designate the executable program files using browse buttons.
For each program, you can select the menu name under which this program will appear in the utility menu.
For each program, you can define if the full path of the sound file currently loaded in Vox Studio is to be transmitted to the external program via the command line.

Input Format command

This command allows selection of the default source conversion format. This is the source format that will originally be selected in the dialog boxes associated with file conversion

operations.
If you plan to convert several files manually from the same format, it makes sense to select this format as the default.

Output Format command

This command allows selection of the default target recording, conversion, and DTMF generation format. This is the target format that will originally be selected by default when you open the dialog boxes associated with file recording, conversion and generation operations.
If you plan to convert several files manually to the same format, it makes sense to select this format as the default.

Directories command

This command allows selection of the default directories that will hold your multimedia sound files and telephony voice files. If you activate "follow changes", then, whenever you navigate to another directory from Vox Studio, this new directory will become the temporary default. If you activate "save on exit" your directory changes will be saved when exiting Vox Studio and become the new permanent defaults.
This command also allows selection of the usual filename extension(s) associated with telephony voice files. Enter the 3-letter extension(s) separated by semi-colons (for instance: vox;spc;mnu).

Batch Conversion Defaults command

You can select the default source directory for files to be converted.
You can select the default target directory for converted files.
You can define the name and location of the batch conversion log file.

Prompter Defaults command

You can define the directory where the recorded files will be saved.
You can select the prompts script file name and location to be used for the next recording session.
You can define the sample rate and resolution to be used for the next recording session.

Group and Ungroup Defaults command

You can select the default directory for the indexed files
You can define the default extension to be used for indexed file names. The ".vap" extension is often used.
You can specify the default maximum number of prompts per indexed file.
You can specify the default sample rate to be used for indexed files. Beware, this has to correspond to the real sample rate used to record or convert the stand-alone files constituting the indexed files.

Miscellaneous Defaults command

This opens a dialog box which allows setting of the following defaults:
DTMF generator tone duration, inter-digit timing, tone amplitude, pause length
Default silence leader length, silence trailer length, silence threshold.

Default target amplitude for normalizations.

Sound Devices command

This command allows selection of one sound input device and one sound output device from the sound devices currently installed under Windows.

Print and return registration card

You can print a dated copy of your registration information and serialization information. This is the first thing you should do after installing Vox Studio on your hard disk.
A fresh-of-the-day registration card printout is required for all tech support and upgrade interventions.

Print Registration command

Print a fresh, dated registration card from the Help menu whenever you need to contact us for support and fax us this document with your request.

Uninstalling Vox Studio

Uninstalling Vox Studio is easy:
Delete all files from your Vox Studio program directory.
Remove the Vox Studio icon from Program Manager.

Menus and commands

File menu

Open command

Opens a file for playback, conversion or filtering. By default Vox Studio looks for files in the directories defined through the Default Directories menu. Vox Studio also looks by default for files with extensions as defined in the Default Directories menu.
If the file you open is a multimedia file (.wav for instance) then Vox Studio gets all the file information it needs from the header portion of the file.
If the file you open is a voice processing telephony file (.vox for instance) then Vox Studio will open a dialog box and request additional information about the file such as sample rate and recording format. This is due to the fact that telephony format files usually do not have a header and the user is the only one able to provide the information. It is, of course, crucial to give Vox Studio the correct file information !

Exit command

Terminates Vox Studio and returns to Program Manager.
You can also double click the top left corner of Vox Studio.

Utilities menu

Program 1 to 4 commands

Sound cards often come with associated programs such as mixers, volume controls and .wav sound editors. These programs, and other useful ones such as, for instance, File Manager can be accessed directly from within Vox Studio.
A maximum of 4 external programs can be accessed through this menu option. Which ones are called is defined through the Default Directories menu.
If one of the external programs is a sound editor you could optionally transfer the name of the current Vox Studio file to the external program's command line so that it would open with the current file already loaded. This saves you a few clicks and you don't have to remember the current file's name. This option is enabled in the Default Directories menu. It will only work if the external program you use does look for a file name on the command line.

Conversion menu

Convert current allows manual conversion of the currently open file.
Batch conversion, as the name implies, allows automated conversion of a whole bunch of files.
The remaining commands allow concatenation or extraction of indexed files.

Effects menu

The commands described below allow you to perform filtering, amplitude normalization and leading or trailing silence adjustments on the current file.
Equivalent commands can be given from the file conversion dialog boxes by selecting option buttons. These operations can then be combined with conversion operations.

Help menu

Contents command in Help menu
Opens Windows' Help program with the help file for Vox Studio opened at the very first topic: Contents of Vox Studio Help.
From there you can navigate to or search for the particular help topic you need.

Search command in Help

The Search command in the Help menu allows you to look for help on specific topics by entering a keyword and searching for all help topics that relate to that keyword.
Don't forget to try the Index and Glossary sections of this help file too.

About Vox Studio command

The About Vox Studio command pops up a box which gives you program version information and serialization information identifying the sole legal owner of this package.
To protect the owner of this package, this serialization information is encoded into the program at various places.

About Xentec command

The About Xentec command pops up a box which gives you a short overview of the company which developed Vox Studio.
You will also find our address, phone and email information there. You can also find the

same address information in this help file under the topic Xentec.
Contact us for orders, technical support, suggestions for new features, insults or compliments. We would love to hear from you.

The Toolbar

The toolbar is a shortcut for the most used commands. The icons from left to right represent the following commands:
File Open, Playback, Record, Prompter, Convert Current, Convert Batch, Help.

File formats

Essentially, Vox Studio knows two generic groups of message file formats: the multimedia formats (usually known as .wav files) and the telephony formats (usually known as .vox files).

Multimedia file formats

The industry standard format for multimedia sound files is the .wav format. These files are usually (but not necessarily) recorded at 11.025 kHz, 22.05 kHz or 44.1 kHz sampling rates with a resolution of either 8 or 16 bits.
In addition to 11, 22 and 44 kHz files Vox Studio can also record and play back .wav files at 6.0, 6.053, 8.0 and 8.117 kHz. These are the sampling frequencies used by most industry standard telephony voice cards.
A nice characteristic of .wav files is that they have a header that contains useful information such as resolution in bits and sampling rate in Hz. Therefore, when Vox Studio reads .wav files it is unnecessary to tell it what the file type is, Vox Studio goes and gets that information from the .wav files themselves (this is not so with telephony type files which usually contain raw data only).
The higher the sampling rate and resolution, the better the sound quality. Unfortunately the storage requirements are also directly proportional to both the sampling rate and resolution:

6,000 Hz Telephone quality, poor
6,053 Hz Telephone quality, poor
8,000 Hz Telephone quality, ok to good
8,117 Hz Telephone quality, ok to good
11,025 Hz AM radio quality
22,050 Hz FM radio quality
44,100 Hz CD quality (if 16 bits)

Telephony file formats

There are several telephony type file formats as enumerated below. The usual file extension is .vox, but this a habit, not a rule. Some voice processing system developers use different file extension names.
Industry standard telephony cards use sampling rates of 6.0, 6.053, 8.0 or 8.117 kHz. For some cards the sampling rate is programmable, for others it is not. The sampling rate to use depends on the card you are using. Consult your telephony hardware supplier for the exact sample rate your voice processing hardware uses.
One of the annoying characteristics of most telephony file formats is that they usually only contain raw data. They have no header with additional information such as coding algorithm, sampling rate or resolution. Therefore when referring to such a file in Vox Studio, or any

other program, it is imperative to specify the exact file coding and sampling rate. This may puzzle you in the beginning, but you will soon learn to discern Vox Studio's difference in behaviour regarding multimedia and telephony files. When Vox Studio reads .wav (multimedia) files there is no need to tell it what the file contains, this information is found in the file itself. When Vox Studio reads .vox (telephony) files it is necessary to tell it what the exact file type is, as this information is NOT in the file. Obviously when Vox Studio has to write in any format it is always necessary to tell it what file type it needs to generate as there is no way for Vox Studio to guess what you wan it to do.
In addition to the formats enumerated below, Vox Studio can concatenate .vox files into .vap format (this operation is called grouping in the program). Inversely, it can ungroup a .vap file into several .vox files.
Here are the various telephony sound file formats known to Vox Studio today:

ADPCM, OKI variant

ADPCM stands for Adaptive Differential Pulse Code Modulation. There are various flavours of ADPCM. The algorithm we have implemented is the original algorithm used by Dialogic voice processing hardware.
It compresses data recorded at 6.0, 6.053, 8.0 or 8.117 kHz sampling rates. Sound is encoded as a succession of 4-bit nibbles glued together in pairs in an 8 bit stream of data. Each 4-bit nibble is essentially representing the difference between the current sampled signal value and the previous value. The compression ratio obtained is relatively modest (12 bits resolution data samples are encoded as 4-bit differentials).
ADPCM coding introduces small signal errors and the sound quality is slightly affected, but it remains sufficient for many telephony applications. Naturally, 8 kHz ADPCM sounds better than 6 kHz ADPCM.
Traditionally, 6 kHz ADPCM is also called 24 kbps (6Kx4) and 8 kHz ADPCM is called 32 kbps (8Kx4).
Not many people know that some cards use 6.0 and 8.0 kHz sampling rates and others use 6.053 and 8.117 kHz rates. Beware when playing back files from one card type onto another. If the files contain voice samples, chances are nobody will ever notice the slight difference in pitch. However, if the files contain frequency sensitive stuff, say DTMF data streams, then the 1.5% difference may in fact turn out to cause severe problems.
Vox Studio has the capability to convert to, and to convert from, indexed ADPCM files as well.

A-law PCM

The European digital telephone network uses a companding algorithm operating on a segmented straight lines approximation to a logarithmic curve called the A-law digital coding standard.
The A-law companders produce 8 bits of companded data per 16-bit sample at a sample rate of 8 kHz. This is also called 64 kbps A-law PCM.
This is the coding algorithm used by PTTs throughout Europe.
Telephony cards capable of recording and playing 64 kbps data produce very good quality voice. In fact you cannot get any better on the current telephone network. Of course, 64 kbps PCM data requires more hard disk space than 24 kbps or 32 kbps ADPCM data, but the voice quality is better. A-law companding produces a better signal-to-noise ratio at low voice amplitudes than Mu-law, but Mu-law has a lower idle channel noise.

Mu-law PCM

The US and Japanese digital telephone networks use a companding algorithm operating on

a segmented straight lines approximation to a logarithmic curve called the Mu-law digital coding standard.
Per channel Mu-law companders produce 8 bits of companded data per 16-bit sample at a sample rate of 8 kHz. This is also called 64 kbps Mu-law PCM.
This is the coding algorithm used in the Bell System throughout the US.
Telephony cards capable of recording and playing 64 kbps data produce very good quality voice. In fact you cannot get any better on the current telephone network. Of course, 64 kbps PCM data requires more hard disk space than 24 kbps or 32 kbps ADPCM data, but the voice quality is better. Mu-law companding produces a lower idle channel noise than A-law, but A-law has a better signal-to-noise ratio at low voice amplitudes.

Linear PCM 8 bits and 16 bits

Linear PCM data is the pure, uncompressed and uncompanded binary code representation of the value of an analogue signal (e.g. voice) after digitization.
Vox Studio can record 8 and 16-bit linear PCM at up to 44,100 samples per second. Of course, this is overkill for most standard telephony applications.
Vox Studio uses a 16-bit representation of signals internally to ensure best possible conversion and filtering results. This is transparent to you, the user, but should indicate how much care is taken of your precious sound samples when we compress, compand, translate and otherwise massage them.
The more bits, the more accurate the signal representation. 8-bit PCM represents signals digitized into as many as 256 discrete step values. 16-bit PCM represents signals digitized into as many as 65,536 discrete step values. The higher the resolution, the more hi-fi your reproduced sound gets. Also, the more samples are taken per second, the better the reproduced sound gets. Obviously, 16-bit linear PCM sampled at 44.1 kHz represents a data stream of 705,600 bps, about 30 times more than 6 kHz ADPCM !

Prompt script file format

The script file is a pure text file. All lines are terminated with the Enter key (Carriage Return / Line Feed pair). The file name is 8 characters long, no extension. The annotation is a short one line description or title. The text for the prompts uses as many lines as necessary and is followed by an empty line (a lone CR/LF pair). The file begins with "Begin Script" and ends with "End Script". Use a text editor to generate or edit script files. Both programs that come with Windows, Notepad and Write (save as text), can produce prompt script files.
Begin Script (CR/LF pair)
(CR/LF pair)
Filename for prompt 1 (no extension here) (CR/LF pair)
Short annotation or title for prompt 1 (CR/LF pair)
Text of prompt 1 on as many lines as needed (CR/LF pairs)
(CR/LF pair)
Filename for prompt 2 (no extension here) (CR/LF pair)
Short annotation or title for prompt 2 (CR/LF pair)
Text of prompt 2 on as many lines as needed (CR/LF pairs)
(CR/LF pair)
etc.
Filename for prompt n (CR/LF pair)
Short annotation or title for prompt n (CR/LF pair)
Text of prompt n on as many lines as needed (CR/LF pairs)
(CR/LF pair)
End Script (CR/LF pair)

Tips and techniques

Sound card quality

You get what you pay for. Although bargains certainly exist, we did find that the sound cards that consistently gave good or excellent results were not the rock-bottom priced, unknown label type. Which multimedia sound card you buy and how much you spend is up to you. If all you want to do is conversions, you do not even need a sound card. However, you need to be warned: there are substantial differences in quality between the various sound cards available on the market today. We still have to see a super cheap one that sounds great. The most important aspects are:
Some multimedia sound cards work very well but only when recording or playing back at the standard sample rates of 11, 22 and 44 kHz. If you also want to play back telephony files at 6 or 8 kHz on your sound card the quality at low sampling rates is important. If you hear superposed distortion that follows the rhythm of the recorded speech you may be hearing aliasing problems. Good products incorporate anti-aliasing filters that work at low frequencies too, cheap clones don't.
A sound card is not a must, you can play files back over a voice processing telephony card !
Select a card that can do 16-bit recording and play back. The 8-bit cards, albeit usable, introduce more quantization noise.
The better cards have lower idle channel noise levels and pick up less stray noise from the PC itself. Some products are so bad that you record hissing noises when the PC's mouse is moved around the desk !
Buy the best sound card you can afford. A few hours of messing-around because of a poor card will cost you more than the difference for a decent sound card. There are lots of very good cards at very affordable prices.
If you are using a cheap sound card and find it to be of impeccable quality, let us now.
Good multimedia cards are usually sold with good accompanying software. Look for a top quality sound mixer and sound editor. Vox Studio does not provide any of the tools that come standard with every decent sound card.

Mixer, editor and volume applets

Vox Studio does not provide a mixer or sound editor for your multimedia sound card. These software applets always come with the sound card itself. Mixers and volume control applets are card-specific. The philosophy behind Vox Studio is to be card-independent. We allow you to access these (or other) proprietary utilities from within Vox Studio itself. Select the programs from the Utility menu.

Clean recordings to start with

To obtain high quality prompts, make sure you start with a spotless master recording. Here are a few common-sense guidelines that will make it easier to obtain good quality "masters". Only spotless recordings convert well and make good prompts.
Be patient and start over until you are absolutely proud of your original master recordings. Never go to the next step until the previous step yields perfect results. Never try to correct imperfect master recordings later using filtering, amplification or conversion manipulations. Always test the quality of your master files before proceeding to convert them.
Use a hyperdirectional microphone to avoid recording surrounding noises and room echo. Do this even if you have a silent studio area.
Use a PC with no fan or with a silent fan. Keep the directional microphone at a distance and directed away from the PC.
Use a quality sound card in a PC with a well filtered power supply. Use a well shielded, low

emission monitor.
If you hear hissing noises in your recordings, which cannot be related to surrounding noise reaching the microphone, swap your sound card for another one. Try another monitor. If that does not help, try swapping your PC platform for one that generates less interference. Make sure no hissing noises are produced while you move the mouse. Once you have found a clean combination of sound card, mouse, monitor and PC, stick with it.
Use a low impedance hyperdirectional microphone and keep leads short, shielded and away from power leads and sockets. Ground your PC system at one point only.
Establish your recording equipment in a furnished mid- sized room. Avoid recording in concert halls and closets. The room's echoes have a palpable influence on sound "quality".
Do not record in a room where others are working. Close your window to keep outside noise away. Avoid recording under the flight path of Concorde. Record in the evenings, at night or during week-ends if you cannot avoid hearing slight workday noises. Even slight noises get recorded and cause problems later.
Stay in the recording room for two minutes, close your eyes and listen with acute attention. Do you hear anything at all ? If you do select another studio.
Make test recordings with silence only. Do this in the recording room, with the PC, sound card and microphone you want to use. Place the mike on the table and start a 30 second silence recording at 16 bits and 44 kHz. Play this recording back through headphones, not through speakers. Listen with great attention. Do you really obtain absolute silence only ? If not, solve this problem before you go one step further. Clean recordings convert very nicely to other formats, recordings with background noises or hissing don't.
Record at reasonable levels. Avoid compensating for low-level recordings by pumping the gain up at a later stage. Avoid saturation while recording.
Keep the microphone 20 cm away from the speaker's lips to minimize "plops" and breathing sounds.
Use the keyboard gently, so as not to record clicks. If you move a mouse on the desk that carries the microphone use a good mousepad. Do not locate the keyboard on the desk that carries the microphone or do use a sound-absorbing mat. Avoid kicking the recording table. An office chair on wheels can generate noise when you move. Test this.
Use experienced speakers with a clear pronunciation and telephone-friendly voice. Perform preliminary tests: some speakers have a propensity to produce DTMF sounds, avoid their services. Test the speakers' voice quality before and after conversion to telephony format and play a few test prompts over the telephone network via a voice telephony card.
A possible procedure to obtain good quality telephony prompts (there are many) would be to use a sound card and Vox Studio to record the prompts as .wav files at, say, 16 bits and 22 kHz, then use Vox Studio to convert those files to one of the telephony types, for instance A-law at 8 kHz.
If you cannot follow above guidelines, have a professional studio record the .wav files for you and use Vox Studio to convert the files to telephony formats and sample frequencies.

Conversion and quality

Converting files from one format to another does not affect sound quality, if all that changes is the data format. Compression, companding and sample rate changes of digitized data are subject to the laws of physics and do cause a slight alteration in sound quality. First you have the degradation resulting from the target format and sample rate itself. Converting a hi-fi 16-bit linear .wav file at 44 kHz into a 4-bit ADPCM file at 6 kHz does, for example, cause a noticeable degradation in sound quality. Then you have the, slight but existing, degradation resulting, essentially, from the re-sampling processes. Record directly into the final target sample rate if your hardware allows it. Always check the sound quality of any conversion process on a typical original before committing hundreds of files to it. Remember that differences are less noticeable when played back over a voice telephony card and to the telephone network. Check quality on the final target hardware.

A clean source file converts much, much better than one with background noise. This may be surprising, but tiny background noises usually end-up amplified in the converted file. Do not spend time tweaking Vox Studio, it does not need it. Spend your time making sure your original recordings are of the highest possible studio quality. High quality recordings convert very nicely, thank you, but junk remains junk. Try the example recordings that came with Vox Studio to benchmark conversion quality. You will see, conversion quality is superb if your original is.

When you down-convert the sample rate of a file you always lose quality because you lose information. These are the laws of physics and we all have to accept them. The resulting quality is not much different than what the file would have been if it were recorded directly in the lower sample rate or resolution format.

Vox Studio can up-convert a low sample-rate, low-resolution telephony file back to, for example, 44 kHz and 16 bits. The resulting sound quality will be similar to the original. You cannot gain bandwidth, and the signal information which was absent in the original will also be absent in the up-converted file.

Vox Studio uses digital signal processing to perform the signal conversions. Each conversion involves many mathematical manipulations of the recorded sounds. Avoid unnecessary conversion steps and record or convert your master file directly to the desired target format.

If you plan to use an external editor to manipulate your files, do this on a copy of your master .wav recording, before conversion to telephony format. Convert to telephony format as the last step in the process.

Sampling frequencies of older cards

Some older Dialogic telephony cards (the D/41-B for instance) use 6,053 Hz and 8,117 Hz sample frequencies.

Newer cards (the D/41-D and D/41-E for instance) use 6,000 Hz and 8,000 Hz.

The difference is not noticeable when playing back speech files from one card to another. However, when precisely calibrated signalling or control tones are recorded the difference may be significant.

Registration and support

It is very important for you to register your copy of Vox Studio as soon as possible. Please have a look at the topics below, and send us your registration printout immediately, if you have not done so yet.

Registration

Registered users, and only registered users, are licensed to use Vox Studio and are eligible for support and upgrades.

Vox Studio requires user information to be entered before you install and use the program. This is required once only, and may have been done by your distributor. You are requested to send us a printout of that information, obtained as indicated below, to register. Once Vox Studio's registration procedure is completed you will be able to proceed with the installation. Vox Studio can be re-installed whenever you need to.

Only one installed version per paid Vox Studio license is allowed to exist at any time. We trace any instance of Vox Studio to protect our legitimate customers, and ourselves. Violations of the software license will always be prosecuted to the maximum extent permissible by law.

You can, at any time, produce a registration printout with the Registration Printout command. Your registration printout is your sole passport to support. Customers who, upon installation, have sent in their fully completed registration printout are eligible for support or upgrades, so do send us your form right away. A fresh-of-the-day registration printout will have to accompany every tech support or upgrade request.

Support

In case of difficulty in using Vox Studio please try and find a solution to your problem in this manual. If this fails contact Xentec for support, we will be happy to be of service. For support, contact us in Belgium at:

Xentec nv-sa
Namenstraat 11
1930 Zaventem
Belgium.

Tel. 32-2-757-0666
Fax 32-2-757-0666
Email 100276.344@compuserve.com

As customers you will get our undivided attention, and we will do our best to help you. Your registration printout is your sole passport to support. Only customers who, upon installation, have sent in their fully completed registration printout are eligible for support and upgrades. Support is given by fax or email. In addition to a complete description of your problem, kindly have the following information ready to be faxed to us:
Today's fresh registration printout (from the Help/Registration Printout menu)
A written description of the command sequence that consistently produces your problem
Windows version number & DOS version number
Complete PC description: CPU, FPU, clock speed, memory size, disk size, disk free
Contents of autoexec.bat & contents of config.sys
Sound card type (if used)

Xentec nv-sa address in Belgium

Xentec

Xentec nv-sa
Namenstraat 11, Nossegem
B-1930 Zaventem
Belgium
Tel +32-2-757-0666
Fax +32-2-757-0666
Email 100276.344@compuserve.com

Vox Studio is developed and supported by Xentec.
Xentec markets and distributes software and hardware building blocks for the telecommunications industry in Europe. Our primary specialization is voice processing. Our customers are value added resellers, systems integrators and OEMs.
Xentec's product portfolio includes application generators, industrial PCs, voice cards and voice processing utilities such as Vox Studio. Write to the above address to be on our mailing list for new products and special offers.
We keep adding to our product line. If you have developed a state-of-the art product, contact Xentec for distribution and support in the European market-place.

Glossary

A-law
The PCM companding standard used in Europe.

AC
Alternating current.

ADPCM
Adaptive differential pulse code modulation. A speech encoding method based on storing only the difference between consecutive speech samples.

Algorithm
Series of well-defined steps or computer instructions to process a signal

Aliasing
A problem causing spurious components in the signal. It occurs when a signal is sampled at a rate lower than twice the highest frequency present in the signal. This results in artefacts at a frequency which is the difference between these highest frequencies and half of the sample rate.

AM
Amplitude modulation. The encoding of information by varying a carrier signal's amplitude.

Amplitude
Distance between high and low points of a signal. There is a direct relationship between waveform amplitude and perceived sound volume.

ASCII
American standard code for information interchange.

Audio
Signals composed of frequencies detected by the human ear, i.e. between 20 Hz and 18,000 Hz.

Audiotex
Voice response service. Dial a number, hear the weather forecast.

Batch
Method where many files are processed automatically, one by one, without any operator intervention

Belgium
A tiny country of 10 million inhabitants. Belgium is bordered by France, the North Sea, The Netherlands, Germany and Luxembourg. Renowned for its chocolate, lace, waffles, beer, good food and Vox Studio. **Xentec** is located in a suburb of Brussels, the capital of Belgium and of the European Union.

Compand
COMpress/exPand, a technique to reduce the dynamic range of a signal and then restore it back to close to its original form

CPU
Central processing unit. The brainy part of your PC.

D/41-B
Voice processing telephony card from Dialogic. D/41-B is a trademark of Dialogic Corporation.

D/41-D
Voice processing telephony card from Dialogic. D/41-D is a trademark of Dialogic Corporation.

D/41-E
Voice processing telephony card from Dialogic. D/41-E is a trademark of Dialogic Corporation.

dB

Decibel. Logarithmic representation of the amplitude of a signal. One decibel is the smallest change in sound volume that the human ear can distinguish.

DC

Direct current.

Decibel

dB. Logarithmic representation of the amplitude of a signal. One decibel is the smallest change in sound volume that the human ear can distinguish.

Dialogic

Provider of components and tools to assemble voice processing systems.

Distortion

Any spurious modification to a sound caused by the process that manipulates it or its digitized representation.

DTMF

Dual tone multi-frequency, also called "Touch-tone" by AT&T. 16 combinations of voice-band tones are used to generate dialling signals. The digits represented are 0-9,*,# and A-D. A-D are not available on standard telephones.

FM

Frequency modulation. The encoding of information by varying a carrier signal's frequency.

FPU

Floating point unit. This arithmetic chip assists the CPU in doing fast calculations.

Frequency

The rate of vibration or oscillation of a signal is measured in hertz (Hz), or cycles per second. The normal human ear can detect sounds ranging from 20 Hz to 18,000 Hz. The telephone network only carries signals with frequencies between about 300 and 3400 Hz.

High-pass

Lets frequencies higher than the cut-off frequency through. Removes the frequencies below the cut-off frequency. There is always a finite slope around the cut-off frequency going from the untouched to the removed section.

Idle channel noise

Residual noise present when the voice signal has a zero amplitude.

Interactive Voice Response

IVR. Dial a number, hear information you select by pressing the keys on your telephone.

IVR

Interactive voice response. Dial a number, hear information you select by pressing the keys on your telephone.

Leading

At the beginning of the sound file.

Linear

Used here with the meaning non-compressed and non-companded, straight representation of the original signal.

Low-pass

Lets frequencies lower than the cut-off frequency through. Removes the frequencies above the cut-off frequency. There is always a finite slope around the cut-off frequency going from the untouched to the removed section.

Milliseconds

Thousandths of a second.

Modulation

Variation of a wave to convey a signal

Mu-law

The PCM companding standard used in the US and Japan.

Nibble

Four bits of binary information. Two nibbles can be stored in one byte.

Normalize
To make the volume of a recording as uniformly loud as possible while minimizing distortion of the sound.

OKI
Provider, amongst other things, of silicon chips that convert an analogue signal into a variant of ADPCM.

PCM
Pulse code modulation. Digital encoding method for sampled voice signals.

Pentium
An Intel CPU chip used in Pcs. Pentium is a trademark of Intel Corporation

Phone banking
Dial a number, find out you are broke. One of the applications for Host Interactive Voice Response (HIVR). Similar to IVR but involves communication and exchange of data with a host mainframe.

RAM
Random access memory. The data your PC manipulates is retrieved from and stored in RAM. Your PC uses RAM chips.

Resolution
The resolution of a recording is indicated by the number of bits used to represent sample values. A 16-bit resolution gives a precision of about 0.003% of full scale. An 8-bit resolution gives a precision of only about 0.8% of full scale value. Use 16 bits if file size and conversion time is not a problem.

Sampling rate
The frequency at which samples of sound are taken.

Signal-to-noise ratio
The ratio of the voice signal amplitude to the noise amplitude, usually expressed in dB.

SoundBlaster
A multimedia sound card. SoundBlaster is a trademark of Creative Technology Ltd

Talk-off
Talk-off is a problem that occurs when a voice signal closely resembles a DTMF tone pair and activates erroneous detection of DTMF digits.

Threshold
Limit of amplitude below which a signal causes no action or detection to take place

Touch-tone
An alternative name for DTMF. Touch-tone is trademark of AT&T.

Trailing
At the end of the sound file.

Vap (.vap)
A usual extension given to indexed voice processing telephony files. Vap files contain several concatenated ADPCM recordings and accompanying annotation text.

Voice mail
Analogous to Electronic Mail, except uses recorded voice messages instead of text messages. Can go from simple multi-user answering device functionality to complex office communication center functionality.

Volume
Loudness of sound signal

Vox (.vox)
A usual extension given to voice processing telephony files. This file extension covers a variety of encoding and sampling rate formats. Vox files do usually lack a header identifying the coding format and sample rate.

Wave (.wav)

A usual extension given to multimedia sound files recorded in Microsoft's standard waveform file format. This file extension covers a variety of encoding and sampling rate formats. Wave files contain a header identifying the coding format, resolution and sample rate.

Windows

Windows is a trademark of Microsoft Corporation