

Vox Studio 3 User Guide



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1.	Contents	5
1.1.	Available documentation formats	5
2.	Introduction	7
2.1.	Introduction to Vox Studio	7
2.2.	The program philosophy	8
3.	Functionality.....	10
3.1.	Recording functionality.....	11
3.2.	Teleprompter functionality	11
3.3.	Tape Loader functionality.....	11
3.4.	File Slicer functionality	11
3.5.	Playback functionality	12
3.6.	Waveform display functionality	12
3.7.	Conversion functionality.....	12
3.8.	Editing Functionality.....	13
3.9.	Multi-prompt file functionality.....	13
3.10.	Offset removal functionality.....	14
3.11.	Sound volume normalization functionality	14
3.12.	Silence trimming functionality	14
3.13.	Intelligibility enhancement functionality	15
3.14.	Filtering functionality.....	15
3.15.	DTMF generation functionality	15
3.16.	DTMF detection functionality	16
3.17.	DTMF filtering functionality	16
4.	Installation.....	17
4.1.	Program Setup	17
4.1.1.	Install the program	17
4.1.2.	Create working folders	18
4.1.3.	Set program defaults.....	18
4.1.4.	Entering your license information	18
4.1.5.	Print and return registration info	19
4.2.	Program Removal.....	20
4.3.	System Requirements.....	20
4.3.1.	Windows.....	21
4.3.2.	CPU and speed	21
4.3.3.	Memory	22
4.3.4.	Display	22
4.3.5.	Disk space.....	22
4.3.6.	Sound card	23
4.3.7.	Microphone.....	23
4.3.8.	Headphone	23
4.3.9.	Speaker	24
4.3.10.	Studio.....	24
5.	Menus, Commands and Buttons.....	25
5.1.	User Interface.....	25
5.1.1.	Main window.....	25
5.1.2.	Minimizing Maximizing Closing.....	28
5.1.3.	Drag and drop load	29
5.1.4.	Drag and drop play	30
5.1.5.	Selecting a sound segment.....	30
5.1.6.	Zooming in and out.....	31
5.1.7.	Scrolling a zoomed graph.....	32
5.2.	File menu.....	32
5.2.1.	New	33
5.2.2.	Open	33
5.2.3.	Close.....	35
5.2.4.	Close All	35

5.2.5.	Save	35
5.2.6.	Save As	35
5.2.7.	Save All	36
5.2.8.	Properties	36
5.2.9.	Exit	38
5.3.	Edit menu	38
5.3.1.	Undo	38
5.3.2.	Cut	39
5.3.3.	Copy	39
5.3.4.	Paste	39
5.3.5.	Delete	39
5.3.6.	Select All	40
5.4.	View menu	40
5.4.1.	Window Limits	40
5.4.2.	Zoom In	41
5.4.3.	Zoom Out	41
5.4.4.	Zoom All	41
5.4.5.	Scroll	41
5.4.6.	Scroll Left	42
5.4.7.	Scroll Right	42
5.4.8.	Previous View	42
5.4.9.	Next View	42
5.5.	Prompters menu	42
5.5.1.	Prompter	43
5.5.2.	Prompter options	44
5.5.3.	Tape Loader	45
5.5.4.	Tape Loader options	46
5.5.5.	File Slicer	47
5.6.	Convert menu	48
5.6.1.	Transform Active	48
5.6.2.	Convert Active	52
5.6.3.	Batch Convert	55
5.6.4.	Group to Dialogic indexed file	59
5.6.5.	Ungroup from Dialogic indexed file	61
5.6.6.	Group to NMS indexed file	62
5.6.7.	Ungroup from NMS indexed file	64
5.7.	Tools menu	65
5.7.1.	Generate	65
5.7.2.	Detect	66
5.7.3.	Monitor Sound Input	67
5.7.4.	Play Selection	70
5.7.5.	Play Window	71
5.7.6.	Play All	71
5.7.7.	Play All But Selection	71
5.7.8.	Pause	72
5.7.9.	Stop	72
5.7.10.	Record	72
5.7.11.	Play a list of files	73
5.7.12.	Options	75
5.7.13.	Defaults menu	78
5.7.14.	Sound Devices	94
5.8.	Window menu	95
5.8.1.	Cascade	95
5.8.2.	Tile Horizontally	96
5.8.3.	Tile Vertically	96
5.9.	Help menu	97
5.9.1.	Contents	97

5.9.2.	Search	97
5.9.3.	License Registration	97
5.9.4.	Print the license	98
5.9.5.	Support.....	98
5.9.6.	About Vox Studio	99
5.10.	Toolbars, buttons, keyboard and mouse	99
5.10.1.	Toolbar buttons	99
5.10.2.	Player/Recorder Buttons.....	100
5.10.3.	Right-click contextual menu	101
5.10.4.	Keyboard shortcuts	101
6.	File Formats	103
6.1.	Script File Format	103
6.2.	Sound File Formats	104
6.2.1.	Bicom format	105
6.2.2.	ITU (CCITT) formats	105
6.2.3.	Centigram formats	106
6.2.4.	Dialogic formats	106
6.2.5.	Elan Informatique format.....	108
6.2.6.	Group 2000 formats.....	108
6.2.7.	IBM Directalk formats.....	108
6.2.8.	IMA ADPCM	108
6.2.9.	InterVoice formats.....	109
6.2.10.	Microlog Intela formats	109
6.2.11.	Microsoft ADPCM.....	109
6.2.12.	Natural Microsystems formats	109
6.2.13.	Newvoice formats.....	110
6.2.14.	Next and Sun formats.....	110
6.2.15.	Nortel Generations formats.....	110
6.2.16.	OKI file formats	111
6.2.17.	Philips VoiceManager formats	111
6.2.18.	PhoneBlaster formats.....	111
6.2.19.	Raw PCM formats	111
6.2.20.	Rockwell formats.....	112
6.2.21.	SCII format.....	112
6.2.22.	SoundDesigner II format.....	112
6.2.23.	Voicetek (Aspect) Generations formats.....	113
6.2.24.	Windows .wav formats.....	113
7.	Tips and Techniques	114
7.1.	Sound card quality	114
7.2.	Mixer editor and volume applets.....	115
7.3.	Clean recordings to start.....	115
7.4.	Conversion and quality	118
7.5.	Sampling frequencies of old cards	119
7.6.	My files do not sound right	119
7.7.	Sample files.....	120
8.	Support section	121
8.1.	Registration	121
8.2.	Support.....	122
8.3.	Functional demo	123
8.4.	Copyright.....	123
8.5.	End-User License Agreement	124
8.6.	Third party trademarks.....	127
9.	Glossary.....	128
9.1.	Glossary	128
10.	Index.....	133

1. Contents

This Vox Studio user guide comprises the following chapters:

Introduction to Vox Studio 3	explains what Vox Studio is
Functionality	details the available features
Installation	shows how to set up the program
Menus and commands	is a complete program reference
File Formats	details supported file types
Tips and techniques	useful tricks and advice
Support section	your license and tech support
Glossary	terminology

We strongly suggest you read the complete user guide before attempting to use the program. This will spare you and us unnecessary correspondence about information readily available in this documentation. If you find this documentation to be incomplete or inaccurate, by all means let us know, and we will improve it.

For additional information and the latest news on Vox Studio point your web browser to the Xentec website at <http://www.xentec.be> and visit the Vox Studio pages.

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1.1. Available documentation formats

The Vox Studio user guide is available both as an on-disk help file "Vox Studio 3.chm" in compiled HTML format and as an on-disk manual "VS3Manual.pdf" in PDF portable document format. Both contain exactly the same information.

The ".chm" help file is best suited for on-line consultation while the ".pdf" file is intended to be printed to your printer for off-line reading as a printed manual. The printable version is directly derived from the help file and may therefore be rather repetitious.

To view and print the ".pdf" file you will need to have the Adobe Acrobat Reader installed. The PDF file format has become a widely accepted standard format for information distribution. It works on most computer platforms. Most systems therefore already have this installed nowadays. If you do not yet have Acrobat Reader, you can obtain a free reader from the following website:

<http://www.adobe.com> or select a special version for your country or language directly from here: <http://www.adobe.com/products/acrobat/alternate.html>

You probably already have all you need to view the ".chm" help file in compiled HTML format. You need to have Internet Explorer version 4.01 or later installed on your computer (this is needed anyway to run Vox Studio 3). You do not have to be using IE though. If you have Windows 98, Me or 2000 you already have all the required files installed. If you only have Windows 95 or NT4 but have Microsoft Office 2000 installed, you also already have all you need to view ".chm" files (the latest Microsoft help files also come in this format). As a result, ".chm" files can be viewed on most newer Windows installations. Please note that the ".chm" format has been announced as the future format for all Microsoft help files and your time invested in updating your system is not likely to be wasted (unless you plan to move away from Windows). There is an interesting article on the Microsoft web site [What](#)

[Is the Minimal Internet Explorer Installation for HTML Help](#) . If you still would rather not install all you need to view help files in ".chm" format, you can use the ".pdf" manual instead, but remember that the ".pdf" file was optimized for printing, not for on-screen reading.

If you require the documentation in yet another file format than those two, contact Xentec with your specific need and we will try and accommodate your request to the best of our ability.

2. Introduction

This chapter introduces the Vox Studio program. It briefly sketches the purpose and basic features of Vox Studio and gives some insight into the approach that led to the development of this specifically telecom-oriented voice product.

[Introduction to Vox Studio 3](#)
[The program philosophy](#)

2.1. Introduction to Vox Studio

Vox Studio is a software tool for telecom professionals. Its purpose is to create speech files for computer-telephony and voice processing applications such as voicemail, interactive voice response, call centers, phone banking, audiotex, telephony portals, voice gateways and the like. Vox Studio speeds up the time-consuming tasks involved in the production of voice files and produces high-quality telephony files. Such files are often classified as "vox" files, hence the name Vox Studio.

With Vox Studio, these voice files (called "prompts" in the industry) can be created on-site using a PC, a multimedia sound card, a decent-quality microphone, and a reasonably-silent recording room.

Vox Studio starts by recording standard Windows ".wav" files. If required, these can be edited in Vox Studio and then converted into telephony files encoded in various flavors of ADPCM, A-law PCM, Mu-law PCM, linear PCM and other formats. In fact, you can convert a message from any supported format to any other supported format. Vox Studio does support a very large portfolio of manufacturer-specific file formats. It can also convert sound files prepared on a Mac into telephony files. This makes the product useful both for telephony system administrators or integrators and for recording studios. The Vox Studio conversion quality is superb and this is one of the reasons why this product has become an industry standard.

Massive prompt-recording sessions (up to thousands of prompts) are facilitated by the "teleprompter", by the "tape loader" and by the "file slicer" functions built into Vox Studio. The prompter flashes texts one by one on the PC's screen while they are read by the voice-talent (the speaker). It is a very easy and productive way of recording a large number of prompts in one session. The tape loader automatically digitizes, cuts and saves a prerecorded studio tape, without operator intervention. The file slicer does the same thing by cutting a large pre-existing ".wav" file into voice "slices". Professional studio recordings can be chopped in pieces very rapidly and saved as separate digital voice files.

Another, highly time-consuming task, automated by Vox Studio, is the cleaning-up (or trimming) of recorded prompts. Vox Studio can add or remove silence at the beginning and at the end of voice files. Vox Studio does this automatically for you and intelligently adjusts the length of silence to your system requirements.

There are a lots of other features built right into Vox Studio, like the reduction of talk-off through DTMF detection and DTMF filtering, automatic power level normalization of all files for maximum intelligibility, and more. Voice processing professionals love to have these capabilities grouped into one easy-to-use and very efficient tool.

Vox Studio 3 is a user-friendly 32-bit Windows application. It is compatible with Windows 95, Windows 98, Windows Me, Windows NT4 and Windows 2000. A sound card is required if files are to be recorded or played. No sound card is necessary if Vox Studio is exclusively used to convert one file

format to another or to do filtering or trimming operations.

Vox Studio is targeted at telephony application developers and their customers. It is also very useful for voice system administrators who need to do regular voice file maintenance. The conversion functions in Vox Studio enable professional recording studios to deliver voice files to their telephony customers even if they know little about the specifics of telephony.

Vox Studio is normally used through a friendly graphical user interface (GUI) but it also offers both a command-line conversion and a DLL conversion capability. This allows programmers to use the Vox Studio sound file conversion features from within their own applications. Command-line and DLL reference information is supplied as a separate file, optionally copied to your hard disk when Vox Studio installs.

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

Vox Studio is designed and marketed by [Xentec in Belgium](#).

[Philosophy behind Vox Studio](#)

2.2. The program philosophy

Productivity

Vox Studio is designed to be a productivity tool for sound engineers who regularly need to produce a very large number of top-quality telecom voice files (but have no time to manually cut, paste and save chunks of sound over and over again). Some of our customers use Vox Studio to produce, literally, tens of thousands of voice files a week. Vox Studio is not another manual sound file editor. There are lots of excellent manual sound editors around if what you are looking for is creativity. If what you are looking for is plain productivity and telecom-specific features, then Vox Studio is what you need.

When you use a traditional sound editor, the amount of time that creeps into the very-repetitive tasks of prompt file recording, cutting, trimming and saving is overwhelming. Most sound editors are good at supporting multimedia file formats, not telecom file formats. This is why Vox Studio's emphasis is on automation, speed and telephony conversions and it does all that on thousands of files with one single command. Basically, Vox Studio does all of the repetitive and boring work automatically but still allows you to do manual tweaking when you want it.

Sound quality

Professional answering services, call centers, voicemail systems, and home banking systems require voice prompts of impeccable quality. Our experience shows (and our users confirm) that the best quality telephony files are obtained by starting from studio-quality recordings which are then converted, using sophisticated algorithms, to the target file format and coding algorithm used by the telephony hardware itself. Direct recording through telephony hardware (and a phone handset) seldom gives acceptable results.

Open approach

Until all hardware and software suppliers agree on a single common file format and coding algorithm for stand-alone or multi-prompt telephony voice files, the diversity of file formats and coding algorithms will continue to be a barrier to development and migration across hardware platforms and software

tools. Vox Studio supports many different file formats from different telecom manufacturers, and we will relentlessly continue to add to our file palette as time goes on.

Let's summarize

If you are a computer-telephony professional and you need to record and then trim, normalize and convert 15,000 telephony prompts for next week then Vox Studio is exactly what you need. If you are an audiophile and want to add reverberation to your latest MP3 live show recording, then Vox Studio is probably not the product you are looking for. The accent is on raw productivity, not on artistic creativity.

Now, let's turn to the next chapters and look at what Vox Studio can do for you.

3. Functionality

This chapter summarizes the main capabilities of Vox Studio. You will find detailed usage information in the chapter on menus and commands.

- Vox Studio provides a graphical user interface (GUI) to record and play prompt files in a wide variety of telecom formats. You need a PC operating Windows and equipped with a multimedia sound card to record or play back.
- Vox Studio can perform file conversions between all the multimedia and telephony file formats it knows. Conversion from stand-alone files to multi-prompt (indexed) files and back is possible too. Vox Studio 3 supports long filenames. You do not need a sound card if you are only going to do conversions.
- The Prompter is similar to the visual tool from which TV news anchors and public speakers read their announcements.
- The Tape Loader will automatically digitize a prerecorded (studio) tape, detect silences between successive prompts, decide where to cut and then save each prompt under a preassigned file name.
- The File Slicer will take a large existing ".wav" file as input, detect the silences between successive prompts, and then decide where to cut and save under a preassigned file name.
- Vox Studio can generate files containing DTMF tone sequences or can insert DTMF tones into existing files. It can also detect and remove DTMF tones from sound files.
- Special tools are available for high-pass and low-pass filtering, DTMF filtering, DC offset removal, silence trimming and volume normalization.
- The above functions can be performed in automated "batch" mode so that you can produce thousands of files very rapidly.
- Vox Studio has drag-and-drop capability. You can select multiple telephony or multimedia files from within another program and drop them onto Vox Studio with your mouse. This will load or play the files you selected. All file formats supported by Vox Studio can be handled like this.
- Vox Studio also offers a command-line interface and a DLL that allow programmers to utilize the Vox Studio voice file formatting, coding and compression conversion capabilities from within their own applications. Detailed command-line and DLL reference information for programmers is available in the on-disk documentation or can be obtained from Xentec.

To find out more about what Vox Studio can do, browse through the subjects below:

[Recording functionality](#)
[Teleprompter functionality](#)
[Tape Loader functionality](#)
[File Slicer functionality](#)
[Playback functionality](#)
[Waveform Display functionality](#)
[Conversion functionality](#)
[Group and Ungroup functionality](#)
[Offset Removal functionality](#)

[Amplitude Normalization functionality](#)
[Silence Adjustment functionality](#)
[Intelligibility Filter](#)
[DTMF Generation functionality](#)
[DTMF Filtering functionality](#)
[High-and Low-Pass filtering functionality](#)

3.1. Recording functionality

Clicking on the buttons of a user-friendly recorder-like device starts recording. Prompts are recorded one by one and can be played back immediately for verification or editing before being saved. For mass recording sessions, where lots of files are involved, you should use the [prompter](#) or the [tape loader](#) functions instead. These are described further in this documentation.

You can easily record 8-bit and 16-bit ".wav" files at preselected frequencies, ranging from 6 KHz to 64 KHz or you can select completely arbitrary frequencies in the same range.

3.2. Teleprompter functionality

The on-screen prompter allows fast recording of lots of multimedia ".wav" files. It uses an ASCII text file script (which you prepare) to flash prompt messages one by one on the PC's screen. Simply tapping the space bar or clicking a mouse button while reading the messages as they appear on the screen records the prompts. The prompts are automatically digitized and saved one by one as .wav files under filenames you have predefined in the script file.

3.3. Tape Loader functionality

The tape loader can automatically digitize a prerecorded studio tape. Connect the output of your tape player to the input of your sound card and start the tape loader in Vox Studio. It will automatically detect where to cut your prompts and will save them all under filenames which you define in a script. It can also work without a script and generate filenames automatically for you. Go for a pizza, and your digitized files will be waiting for you when you come back.

The tape loader can even be used as an accelerated voice-operated prompter as it recognizes silences between spoken messages. It will display the text of the next message on-screen as soon as it detects the end of the previous one. All you have to do is read the messages into a microphone.

3.4. File Slicer functionality

The file slicer can automatically slice a long prerecorded ".wav" file into individual prompt files. It will

automatically detect where to cut your prompts (it looks for silences) and save all the prompts under filenames which you define in a script. It can even work without a script and generate filenames automatically for you. If your professional recording studio delivers all your prompts stuffed into one large ".wav" file, this is an easy way of cutting it all into separate prompts.

3.5. Playback functionality

Any file loaded in Vox Studio can be played using the Play, Pause, Fast Forward, Fast Reverse and Stop buttons conveniently located in a toolbar at the bottom of Vox Studio's main window. The fast Forward and Reverse buttons don't just move your position in the sound file, they actually play the file at high speed in forward and reverse mode! This may sound strange but is actually very useful if you have a transcription job to do.

There is also a Play List command, which allows simple playback of a whole list of files. These files can be ".wav" files or telephony files (ADPCM, A-law, mu-law, you name it).

Drag-and-drop playback capability is provided as well. Multiple files can be dragged from an external program and dropped onto Vox Studio to start the Play List command.

3.6. Waveform display functionality

Vox Studio displays the waveform of each loaded sound file in a separate graphical window. The waveform display is essential when you need to manually intervene and do some cutting and pasting in sound files. The waveform display also gives valuable feedback on problem situations like files that were recorded at too high or too low amplitude. Noisy background sounds are also clearly visible. Naturally, the graphical display allows zooming and scrolling.

Two generic display modes are available: you can look at the linear waveform amplitude or you can choose to work in dB mode where the power levels of the sound files are shown on-screen in decibels.

Last, but not least, there is also an input monitor which does not display the content of a file, but rather shows the signal present at the sound card input. This makes it easy to adjust the sensitivity of your recording setup. The monitor has three display modes: waveform, VU-meter, and spectral display.

3.7. Conversion functionality

Vox Studio converts any file format it knows to any other file format it knows. This includes the capability to change the sampling rate (downwards and upwards) as well as the coding or compression algorithm.

You can, for instance, convert a 44.1 KHz ".wav" file into a 6 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 voicemail file into a 22 KHz ".wav" multimedia file. This would, for instance, allow you to review voicemail files on a normal PC running Windows.

The amplitude of the recorded signals is left unchanged by the conversion processes. You can, however, elect to activate the "Normalize Sound Volume" option during conversion. This gives you the ability to do post-recording amplitude adjustments. You can also apply a variety of filters during conversion. Conversion and filtering of files can be done in one step. Vox Studio does all the filtering that is necessary for sample rate conversion transparently and you do not need to apply any pre or post-conversion filters.

See the chapter on [Sound File Formats](#) for a complete summary of what files Vox Studio can record, play back, convert or otherwise manipulate.

See the [Convert menu](#) chapter for more information on how to convert files.

3.8. Editing Functionality

Sound files loaded in Vox Studio can be manually edited. You can zoom-in on part of a signal, select a portion of the sound signal, then make a copy of it to the clipboard, cut it to the clipboard or simply delete it. The clipboard content can then be pasted again into the same or another sound signal within Vox Studio. With the editing capability, missing words can be added, superfluous noises can be cut out and sound files can be combined.

Read more about editing in the [Edit menu](#) chapter.

3.9. Multi-prompt file functionality

Some telephony hardware manufacturers promote the use of multi-prompt files. These files contain more than one message. Such files are also called "indexed files". It is therefore important to be able to create indexed files and to modify single messages within the indexed files.

You can load an indexed file in Vox Studio, then view or edit the content of each separate message contained in the indexed file and then save the indexed file back to disk. Indexed files appear in the file tree panel as one icon for the indexed file with as many sub-icons as there are messages in the indexed file. When you click a message icon you see it in the graph panel and can edit it there. This is a very useful productivity feature.

Vox Studio also offers the ability to group several individual messages into a single indexed file or to ungroup an indexed file into its separate components.

Grouping: Any number of individual, stand-alone prompt files can be converted into one, single, larger indexed file. A script file directs this automated procedure. The 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. A script file can optionally be produced. This script file is then compatible with the prompter script file and the group script file described above. It is thus possible to ungroup, then regroup an indexed file without having to create a script file first.

Thanks to this script file technique it is possible to rapidly record individual prompts with the prompter,

group them into an indexed file, ungroup the files, rerecord one of the prompts then regroup it all to an indexed file, etc.

Vox Studio currently supports loading, editing, grouping and ungrouping for native Dialogic (vap) and NMS (vox) indexed files. You could even ungroup a file in one indexed format into its separate components, do a format conversion on the components and then regroup these in the other indexed format.

[Group to indexed Dialogic](#)

[Ungroup from indexed Dialogic](#)

[Group to indexed NMS](#)

[Ungroup from indexed NMS](#)

3.10. Offset removal 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.

Some low-cost sound cards can introduce very significant and disturbing DC offsets. You don't necessarily hear the offset, but it may cause a loud click at the beginning or end of a file. Centering removes the signal's DC bias, and is therefore also useful if the signal has to be amplified considerably.

3.11. Sound volume normalization functionality

For intelligibility and consistency purposes, menu prompts and recorded messages should all be played loud and clear to the telephone line, and if possible they should all have similar perceptible sound volumes.

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. Sometimes it may be necessary to combine messages recorded during different recording sessions. These messages may not all have the same sound level.

Here again Vox Studio can help. Vox Studio can automatically normalize the sound energy in all your files. Define the desired maximum sound energy, then sit back and relax while Vox Studio does the dirty work for you.

3.12. Silence trimming functionality

Voice processing professionals often use a voice editor to 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. If you have to do this on thousands of files, it is a nightmare.

You can completely automate this repetitive task with the help of Vox Studio. Simply select a default length for leading and trailing silence in the program defaults and give a single command to apply it to all your prompts. Vox Studio does all the work for you. This works on multiple files in batch-mode too. You can do this on thousands of files in one go!

Automatic silence adjustment is a threshold-activated process and, naturally, requires very clean recordings.

3.13. Intelligibility enhancement functionality

The Intelligibility Filter capability consists of a "Clarity" filter which corrects the muffled sound effect obtained when down-sampling voice files and a "Boost" option which produces variable signal amplification in order to increase the perceived voice volume (energy) content of recorded files.

The Intelligibility Filter is an option which can be selected while doing any of the conversion operations.

The Intelligibility Filter parameters can be set in the program defaults. A "weak" clarity filter and "no boost" is usually the best choice, but it may require some trial and error to figure out what setting is best for your sound files.

3.14. Filtering functionality

The high-pass (or if you prefer low-cut) filter has a programmable cutoff frequency. It effectively removes frequency components below a selected cutoff frequency from your sound file. This is useful to remove rumble and power line noise.

The low-pass (or if you prefer high-cut) filter also has a programmable cutoff frequency. It effectively removes frequency components above a selected frequency from your sound file. This is useful to remove hiss.

Use these filters only when needed and if you know precisely what you are doing. This is going to affect your digitized files. If Nyquist or aliasing mean nothing to you, you should probably not touch these functions. Always test filtering and other effects you apply to sound files before you get rid of the valuable originals.

There is NO need to use these filters when you do sample rate conversions as Vox Studio applies anti-aliasing filters automatically and transparently.

3.15. DTMF generation functionality

Complex DTMF (touch-tone) sequences of various lengths can be added to sound files. DTMF digits and pauses are typed on a graphical telephone keypad (like a portable dialer). The DTMF digits 0 to 9, A to D and * and # are all available. The default length of the individually produced DTMF tones and

pauses is settable in the program defaults.

3.16. DTMF detection functionality

Complex DTMF (touch-tone) sequences of various lengths can be detected in sound files. The DTMF digits 0 to 9, A to D and * and # are all detected. The threshold amplitude, minimum duration and maximum gap length are settable in the program defaults.

3.17. DTMF filtering functionality

Some recorded files with human voice or music can repeatedly cause 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, contains pairs of frequencies very similar to the DTMF frequencies generated by telephone keypads. This typically happens only once in a while, and only on a small number of files.

Vox Studio can fix this for you by removing the 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 somewhat. 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 a live system. Vox Studio's DTMF detection capability allows automatic flagging of files that may contain such erroneous DTMF tones. There is absolutely no justification to filter all your files for DTMF.

Vox Studio provides for selection of weak, medium or strong DTMF filtering.

4. Installation

Read the following system requirements and setup procedures before installing Vox Studio on your PC.

[Program set-up](#)
[Program removal](#)
[System requirements](#)

4.1. Program Setup

As well as installing the software with the supplied setup program, you should also create a custom working environment for Vox Studio so that you can work efficiently. Source and target directories should be created for your files and your program preferences should be set using the Defaults menu.

[Install the program](#)
[Create working directories](#)
[Set program defaults](#)
[Entering your license information](#)
[Print and return registration form](#)

4.1.1. Install the program

Insert your CD in the CD player and close the CD tray. The setup program should start automatically. If it does not, because your PC settings are inhibiting this, just start the setup program in the root directory on the CD. If you downloaded Vox Studio from a web site or received it by e-mail, just double click on the downloaded file to start installation. If you did get Vox Studio as zip archive you should double click the file "setup.exe" to start installation. The installation process copies all necessary files to the Vox Studio folder of your choice. Additional sample files may be left on the installation disc for you to retrieve later if you so desire.

You will be given the choice to install the Vox Studio program (the normal choice for average users) and/or additional documentation which programmers can use if they wish to use Vox Studio from within their own programs (for advanced programmers only). Vox Studio does not install or overwrite any system files during installation. All the files Vox Studio installs go into the Vox Studio program folder.

Installation should be very straightforward but if you nevertheless have any problem installing Vox Studio, and you cannot find the cause of the problem yourself, contact Xentec by e-mail at support@xentec.be or call us directly at +32 2 757-0666 (during European working hours).

For your safety, we perform thorough virus checks on all the programs we make available on our web site and on the ones we upload to other sites ourselves. We cannot, however, check all the other sources through which Vox Studio may be made available. It may therefore be wise to scan the installation package for possible virus presence, especially if you obtained it from a source that does not guarantee the safety of its downloads.

4.1.2. Create working folders

You will most probably use a lot of folders (directories, for the technically inclined) in order to keep your sound and program files well organized. We suggest you plan on having at least three directories:

- One of them will hold Vox Studio and associated program files. You select this location at installation time.
- The second could be the default directory to hold the multimedia source files (in ".wav" format for example).
- The third could be the default directory to hold the target telephony files (often, but not always, with a vox extension).

4.1.3. Set program defaults

You can set all Vox Studio defaults from the Tools/Defaults menu. Please read the next sentence very carefully. **At the very least you should set the default folders, file extensions and sound devices or Vox Studio will not work satisfactorily.**

Once you get used to working with Vox Studio you should set all the other defaults too. If you set appropriate defaults in the Tools/Defaults menu you will greatly reduce the amount of repetitive manual intervention required in your daily work.

[The Tools/Defaults menu](#)

4.1.4. Entering your license information

The first time Vox Studio is run it will open this type of dialog box:

Vox Studio registration

Please fill-in all identification fields below

Name:

Company:

Street:

City:

State:

Zip code: Country:

Phone: Fax:

To register, fill-in the License ID and associated License Key below. Registered copies of Vox Studio are linked to a License ID which identifies you uniquely. Every License ID is tracked.

You are therefore responsible for ensuring correct usage of this program, according to the terms of the license agreement. Licenses and Keys cannot be returned and you remain responsible for their correct usage. In case of doubt click <Cancel> now.

License ID:

License Key:

Please enter the complete information in all fields. If you are using Vox Studio in demo mode, and have not purchased a license, leave the License ID and License Key fields as they are and Vox Studio will run in evaluation mode. If you purchased a license enter your personal licensing information in those fields. Please make certain that you copy the personal ID and Key information exactly. Take care with capitalization and make sure you do not substitute O (upper case oh) for 0 (zero) or 1 (one) and I (upper case i) for l (lower case L). If you can, copy and paste this licensing information into the fields from a pure text file. Do not paste this from a word processor, as you are likely to introduce invisible formatting characters in doing so.

4.1.5. Print and return registration info

You can print a dated copy of your registration and serialization information from the Help menu. This contains all the information you may need to obtain technical support on the product or enquire about an upgrade.

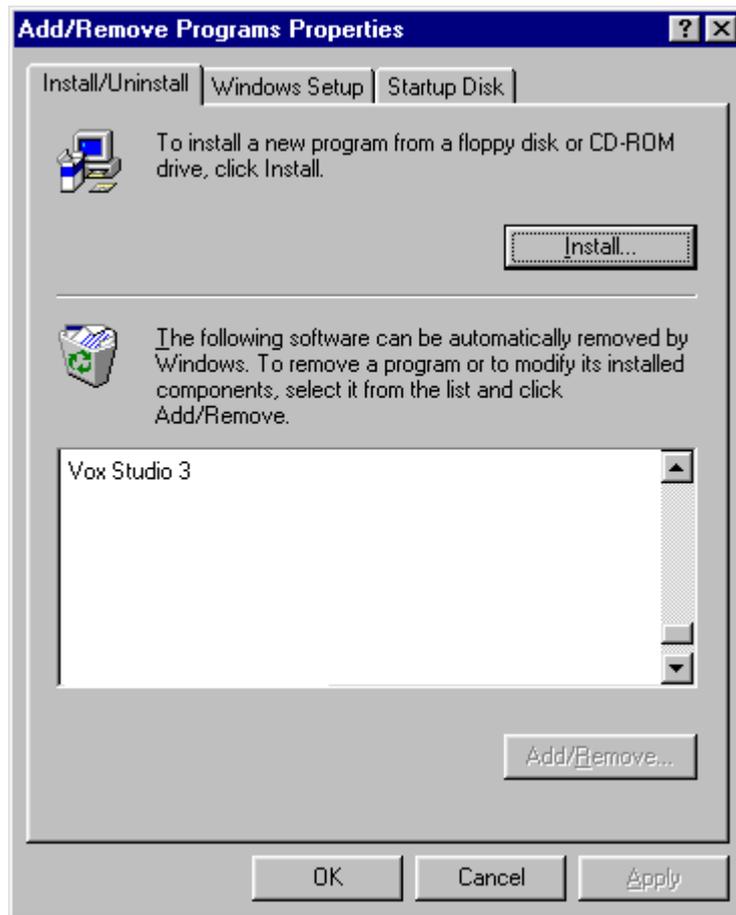
Demo-mode users can fax us their registration printout to be included in our future mailings. This is optional, of course.

[Print the license](#)

4.2. Program Removal

Uninstalling Vox Studio is easy:

From Windows click Start/Settings/Control Panel/Add-Remove Programs. In the list select Vox Studio and click on Add/Remove.



This will remove Vox Studio but will not remove the other settings, files, folders and preferences you created after installing Vox Studio. If you want to remove these too, delete all files from your Vox Studio program directory and its subfolders (if any). Also remove the subdirectories containing your sound files if you no longer need them.

4.3. System Requirements

What system you need is (partially) a function of what you want to achieve. Read the following guidelines to help you decide on which kind of PC you are going to install Vox Studio:

[Windows requirements](#)

[CPU and speed requirements](#)

[Memory requirements](#)

[Display requirements](#)
[Disk space requirements](#)
[Sound card requirements](#)
[Microphone requirements](#)
[Headphone requirements](#)
[Speaker requirements](#)
[Studio requirements](#)

In addition, before you purchase a Vox Studio license you should run the free evaluation version of Vox Studio on the same PC, just to make sure there are no unexpected incompatibilities.

4.3.1. Windows

Vox Studio 3 is a 32-bit Windows application. It was written to be used with Windows 95, Windows 95 OSR2, Windows 98, Windows 98 SE, Windows Me, Windows NT4 Service Pack 6 and Windows 2000. It may, or may not, run on Apple computers with Windows emulation; we have not tested this ourselves and therefore do not guarantee functionality on these machines.

Obviously, you need a PC equipped with a 32-bit version of Windows to use Vox Studio 3. You need to have Windows sound card drivers properly 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.

The toolbar functionality and the Vox Studio help file require at least Internet Explorer version 4.01 to be installed on your system. This functionality is tightly integrated with the operating system. If you use an old operating system such as Windows 95 you need to make sure to have at least version 4.01 of Internet Explorer installed. If not, IE can easily be downloaded and installed from many web sites. May we suggest you install IE 5.5 or later? All you need is a minimum installation. It does not have to be your main browser, neither do you need a web browser, you just need the operating system functionality that comes with IE 4.01 and later.

4.3.2. CPU and speed

Most of 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 filter operation. This is done using the PC's CPU. No additional hardware is required. Vox Studio really shines by the quality of its conversions and by the amazing speed at which they are performed.

A 100MHz Pentium is probably the lowest acceptable platform for Vox Studio off-line conversions, but it is the bare minimum and may be too slow for real time operations on file types that require complex algorithms. Of course, the faster your PC, the better. A 400 MHz Pentium or better is the ideal platform for Vox Studio to really fly. Vox Studio is so optimized that most filtering operations are performed faster than real-time.

If your PC is very slow, you will still be perfectly able to perform any conversion you want. You may, however, hear pauses (jerky interruptions) when playing back some telephony format files on that PC's sound card. This is simply because Vox Studio decompresses the encoded files on the fly and

these decompression computations will take more time on that PC. The machine needs to stop playing sound to allow your CPU to catch up with its number-crunching. The more complex the compression algorithm, the more likely this is to happen. In summary, speed is critical only if you want real-time playback of compressed files or complicated recording operations, not for conversion. As the computational speed of PC increases with each PC generation, this becomes less and less of an issue.

4.3.3. Memory

Vox Studio does not rely exclusively on available memory. If it did, it would be impossible to manipulate very large voice files like, for instance, 30-minute recordings. Vox Studio uses your hard disk and temporary files as a scratchpad area. This is invisible to the user, except for the obvious disk space requirement.

Still, for Windows to run comfortably without too much disk swapping, we recommend a minimum of 32 Mbytes of RAM, and 64 Mbytes or more would be a better idea. You can get it to run with less, but you will not be very happy with less. This is really a Windows performance requirement, not a specific Vox Studio need.

4.3.4. Display

Vox Studio version 3 requires at least a 800 x 600 display setting with a minimum of 256 colors. The program screens are designed for use with the standard small-character fonts.

We do not guarantee full functionality with odd display configurations and special drivers, but we really do not expect problems.

4.3.5. Disk space

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 the original voice files and the conversion results plus some temporary scratchpad 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! It is up to you to make sure there is enough space available on the hard disk used.

Vox Studio uses your "Temp" directory as a scratchpad area. It is therefore important that this temp directory resides on the local disc (partition) with the largest amount of free space and that there is enough free space on it to hold all the work files and temporary files. Remember, the hard disk partition that hosts your system's "Temp" folder may not be the same as your working partition. Vox Studio uses the standard Temp of Windows or the folder you have selected in Tools/Options/System. If you want to find out what the standard Windows Temp is, just open this dialog box in

Tools/Options/System and press "Use Windows temporary folder".

4.3.6. Sound card

You need a multimedia-compliant sound card if you want to use Vox Studio to record or play back. No sound card is needed at all if Vox Studio is used to perform format or coding conversions or filtering only.

A sound card is not a must for playing back files if you have another setup outside of Vox Studio to play files: over your voice-processing telephony card for instance. You may not be able to play back ".wav" files that way unless your telephony card driver allows it, and the quality may be no better than telephony quality. Vox Studio drives multimedia-compliant sound cards, but not telephony voice cards.

You should select a sound card that can do 16-bit recordings and can handle sample frequencies of 44 KHz (or more) down to 6 KHz, and do it well. Your sound card should also be able to record or playback mono signals at all the sampling frequencies of interest to you. Most cards perform fine at the standard 11 KHz, 22 KHz and 44 KHz sampling frequencies. However, where you can often really see a big difference, is how they perform at nonstandard frequencies like 6 KHz and 8 KHz or 48KHz and 64KHz. Select a sound card that is immune to the PC's power supply electrical noise.

By all means, select a card that comes from a reputable vendor that supplies quality drivers and driver upgrades, a Windows sound mixer, level adjuster and some limited graphical sound editor (to test the card's hardware). Most brand name products do.

4.3.7. Microphone

A directional microphone is the best companion to Vox Studio. It helps reduce stray noises from the PC's power supply fan, coughing colleagues in the office, passing airplanes, the office photocopier or fax, 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.

Make sure the output impedance of your microphone is matched to the input impedance of your sound card. Also, in case your sound card has both a line input and a microphone input, make sure you connect the microphone to the correct input. Typically line inputs expect a high AC input voltage from a high impedance device. Microphone inputs, on the contrary, expect a low AC input voltage.

Try your microphone with the built-in recording capability of Windows before you purchase a Vox Studio license.

4.3.8. Headphone

Although this is not actually a requirement (sound files can be played over your usual multimedia speakers), headphones are perfect to evaluate the quality of files you have just recorded or converted. Watch for those tiny background noises and slight breathing or paper-shuffling sounds. Of course, if

you work on sound files and you are not alone in your office, a set of headphones might actually be a practical requirement.

4.3.9. Speaker

We mean the "human" speaker here, not the loudspeaker. Recording studios call them "voice talents". This is probably the most impalpable parameter of all, but it is also one with a huge influence on the ultimate result.

Experience shows that just as there are "radio voices", there is something like a "telephony voice." Some people have the ability to make themselves more clearly understood over one medium than over the other. A good radio voice doesn't necessarily make a good telephony voice and vice versa.

Most telephony cards work at 6 KHz or 8 KHz sample rates. This means that the highest voice frequency component these cards can actually play back to the phone line never exceeds 3 or 4 KHz. Worse, the telephone system itself limits the bandwidth to 3.4 KHz. It is therefore usually better not to select a voice talent with a lot of high-frequency content: a lot of what they pronounce would get lost in the process anyway. This is detrimental to intelligibility.

4.3.10. Studio

The main characteristics to look for are:

- Absolute silence
- Room geometry

You may have to experiment to find a suitable location. Try to find a reasonably soundproof area in your office so that outside noises and PC-hum is eliminated entirely (or at least drastically minimized). All spurious sounds will otherwise end up recorded in your prompts.

The size and geometry of the recording room, and the way it is furnished, have an important impact on the perceived quality of your recordings. The resulting impression can go from cathedral-like reverberation to a "locked-up-in-a-wardrobe" kind of sound. Find a reasonable compromise. If you cannot find a suitable studio your microphone will have to be extremely directional. If you cannot afford one of those, get the recording work done by a professional studio. For more information, look in the chapter on [tips and techniques](#).

5. Menus, Commands and Buttons

All Vox Studio commands can be activated from the menu bar at the top of the main Vox Studio window:



Some handy command shortcuts are also available through toolbars and buttons as shown in the [user interface](#) section.

Note to program developers: You can incorporate the Vox Studio conversion capabilities into your own products. All file conversion commands can also be given from external programs using the command-line API or the conversion DLL API. Contact Xentec or browse our website for detailed programming information.

[User Interface](#)

[File menu](#)

[Edit menu](#)

[View menu](#)

[Prompters menu](#)

[Convert menu](#)

[Tools menu](#)

[Defaults menu](#)

[Window menu](#)

[Help menu](#)

[Buttons, mouse and keyboard](#)

5.1. User Interface

The User Interface chapter describes:

The [Main Window](#) which controls Vox Studio

[Maximizing and Minimizing windows](#)

[Dragging files to Vox Studio to load them for editing or conversion](#)

[Dragging files to Vox Studio to play them through your sound card](#)

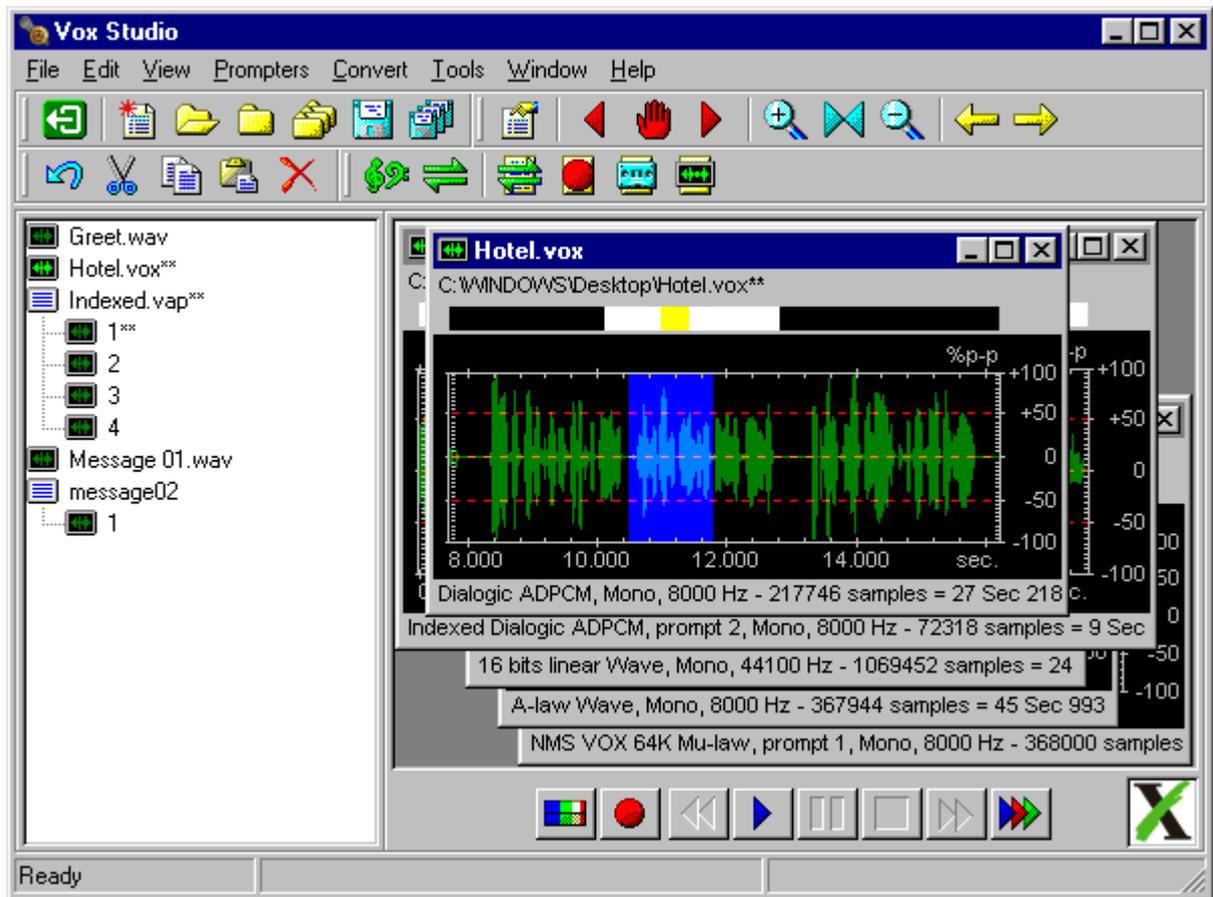
[Selecting a sound segment](#)

[Zooming in and out](#)

[Scrolling in a zoomed graph](#)

5.1.1. Main window

The main Vox Studio window looks like this:



Most program and graph windows (including the main window of Vox Studio itself) can be resized by holding the cursor over the bottom right corner of the window (until a double arrow appears) and then clicking and dragging until the window has the desired size. Double clicking the colored program title bar at the very top is a simple trick to rapidly maximize the size of the main Vox Studio window. Another double click on the title bar will restore the window to its previous size.

All commands can be activated from the menu bar at the top of the main Vox Studio window. The menu bar is located at the top of the screen underneath the program's title bar. It has the menu commands "File", "Edit", "View" and so on. You can activate the menu commands by clicking on any of the menu items. This will then open a list of actions from which you can select the one you want to perform. The menu bar simply looks like this:

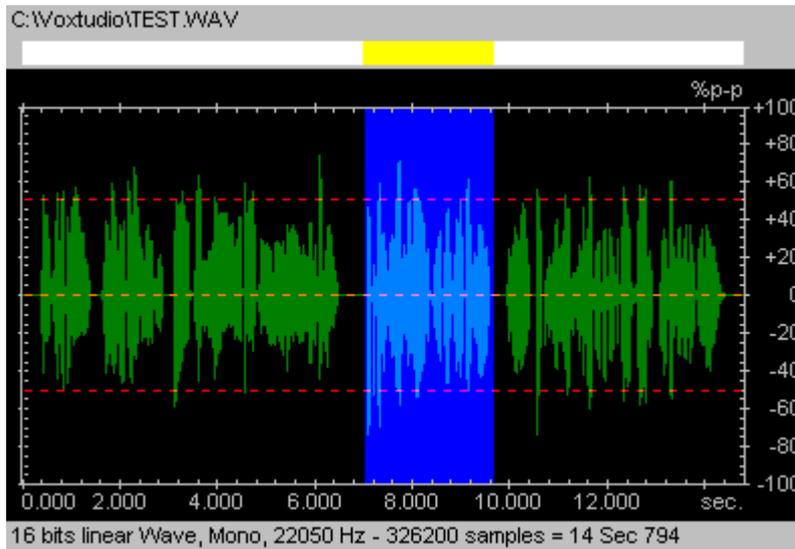


Under the menu bar you will find toolbars that have one-click buttons for most of the important commands, but not for all of them. The toolbars look like this:

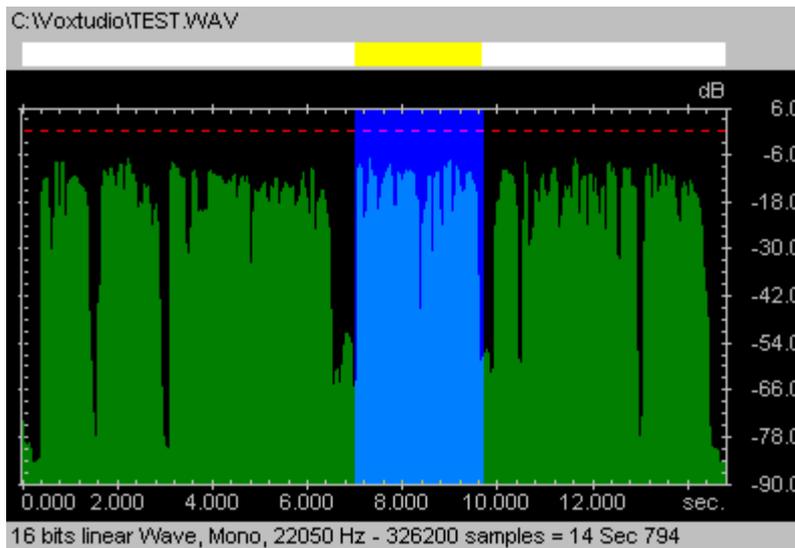


In the beginning, while you are still learning Vox Studio, you can rest your mouse cursor over the toolbar buttons for about half a second and a yellow ScreenTip will appear to explain what the button does.

The graph pane on the right, under the toolbars, is where the currently loaded files are shown as waveforms, each in its own graphical window. The files can be edited in the graphical windows: you can select, copy, cut, delete and insert sound chunks. Each of the graphical windows has a status bar at the bottom that shows detailed information on the file's type, encoding, sampling rate, number of samples and length in seconds. You can change the default display font for the horizontal and vertical scales by setting it in the Tools/Options/Graph fonts dialog box. You can even show or hide various elements of the graph window such as the position bar on top, the scales, the sound file format and other visible items by configuring this in the Tools/Options/Layout dialog box.



The sound signal can also be displayed as energy expressed in decibels, it then looks like this:

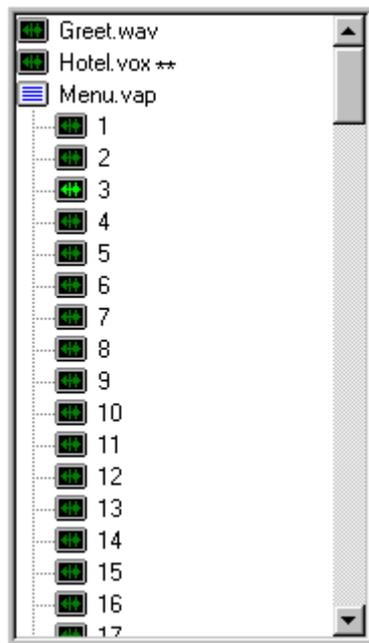


Under the graph pane you will find buttons that make it possible to record, rewind, play, pause, stop or fast forward, very much like a cassette recorder. This is what the player buttons look like:



The file tree pane, located on the left of the main window under the toolbars, shows the names and

icons of all the files you have loaded in Vox Studio. When you click one of those file icons the corresponding waveform is shown in a graphical window in the graph pane on the right. When a filename has ** appended to it, this signals that the image in memory has been changed from the original disk image (the sound signal has been edited, filtered or converted in Vox Studio).



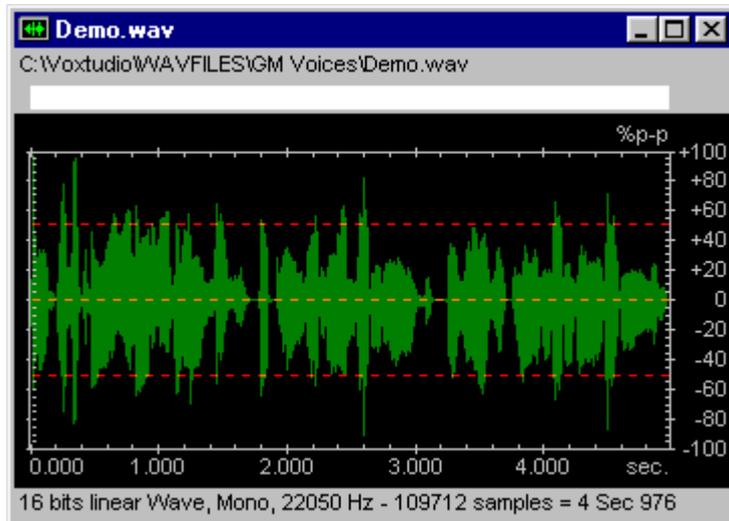
There is also a general status bar at the bottom of the main window. It tells you what Vox Studio is busy doing and also gives useful information on the cursor position in a file while you view or edit a file.

All commands and dialog-box selections are normally executed by clicking the left mouse button, but they can also be given from the keyboard. Each menu command or dialog box selection is shown on-screen with one of the characters underlined. The underlined character is the keyboard shortcut for that command.

Note to program developers: all file conversion commands can also be given from external programs using the command-line API or a conversion DLL API. Contact Xentec for detailed programming information.

5.1.2. Minimizing Maximizing Closing

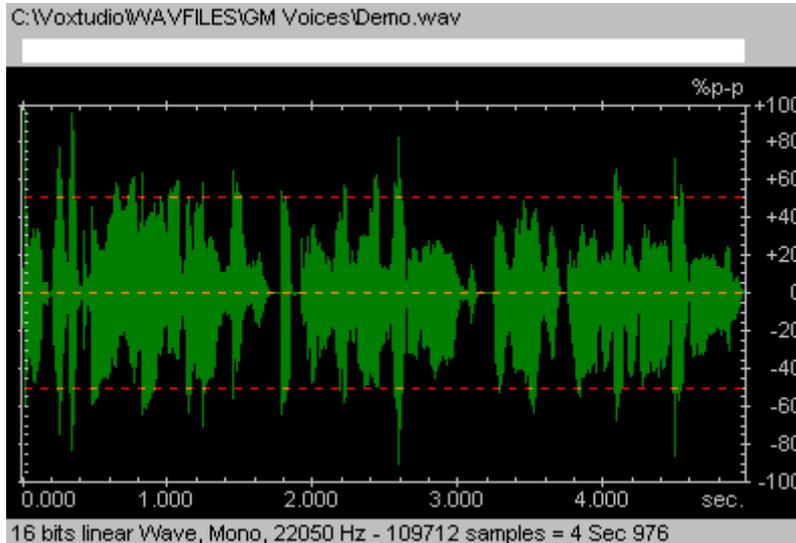
The traditional window control buttons (minimize, maximize, restore and close) look like this  they are located in the top right corner of a window and they allow you to change the size of the graphical waveform windows shown in the graph pane. A normal window looks roughly like this:



If you minimize a graph window it becomes tiny and hides at the bottom of the graph pane (but you can move it around) and it takes no more space than this:



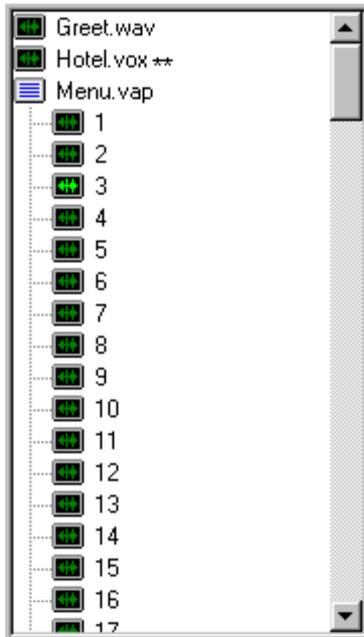
If, instead, you maximize a graph window it will fill the complete graph pane, thereby temporarily hiding other graph windows shown in the graph pane, and its window control buttons will seem to disappear (they don't disappear, they just migrate to the right of the Vox Studio menu bar). A maximized graph window looks like this:



5.1.3. Drag and drop load

Open Windows Explorer (or an equivalent program of your choice), select a folder and, using Windows' usual Left Click, Shift-Left Click, Control-Left Click mouse commands, select all the files you want to load in Vox Studio. Drag the selection over the screen and drop it over the Vox Studio window

(or, if Vox Studio is minimized, hold it over the icon in the taskbar until the Vox Studio window opens, then drop it), et voilà, your files will all load in Vox Studio. That works not only with ".wav" files but also with telephony files. You cannot mix file types within a single drag and drop operation though; all files should be of the same type. For simplicity it is advisable to drop the selected files over the file tree pane as dropping them elsewhere may actually trigger the playback of the dropped files (see next paragraph).



Drop the files in the window to load them.

5.1.4. Drag and drop play

Open Windows Explorer (or an equivalent program of your choice), select a folder and, using Windows' usual Left Click, Shift-Left Click, Control-Left Click mouse commands, select all the files you want to play in Vox Studio. Drag the selection over the screen and bring it over the Vox Studio window (or, if Vox Studio is minimized, hold it over the icon in the taskbar until the Vox Studio window opens) and then drop it on the rightmost button of the player toolbar (the button with blue, red and green arrows), et voilà, your files will all play in Vox Studio. This works with ".wav" files but also with telephony files. You cannot mix file types within a single drag and drop operation though; all files should be of the same type.

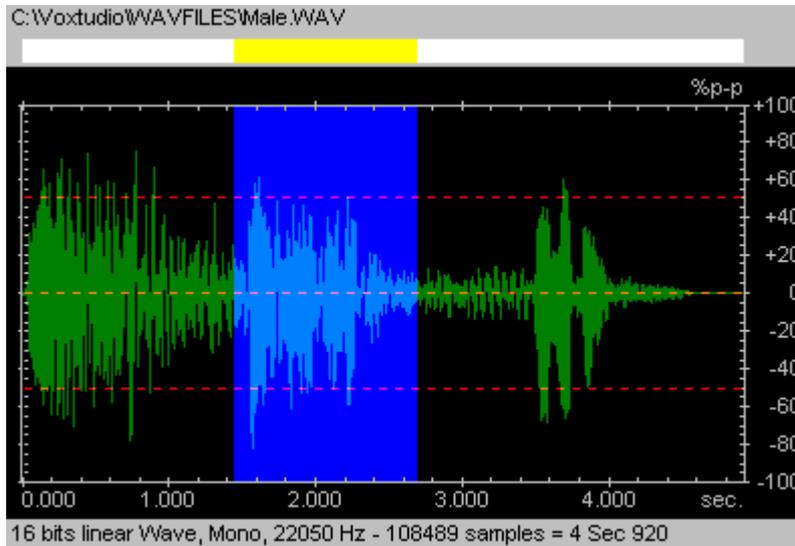


Drop the files on the rightmost  button to play them.

5.1.5. Selecting a sound segment

Selecting a sound segment is simple: just click and drag the mouse over the area you want to select.

The selected area changes color. You can enlarge or shrink an existing selection by pressing the Shift key on your keyboard and clicking with your mouse to the right or left of the current selection. The selection can be extended or reduced. You can listen to the current selection by pressing the play button.



You can see what relative part of the sound file has been selected, even if the file has been zoomed-in. The current selection shows in yellow in the position bar on top of the graph window. If the file has been zoomed-in then the position bar still shows the selected segment, but it may now look like this:



The white part is the visible window, the black part is the sound that is currently not shown in the window, the yellow part is the current selection, if any.

To remove a selection click outside the graph area near the graph borders (where the scales are).

If a sound section is selected the section will play when pressing the play button and the selection will be edited when doing a copy, cut, delete or paste operation.

If you just click once inside a graph (without dragging the cursor) then only a vertical cursor line will be placed on the graph. This is the starting point for a play operation or the insertion point when you do a paste operation.

If there is neither a selection nor a cursor then the whole file will play when you press the play button and any pasted sound will be inserted at the end of the file.

5.1.6. Zooming in and out

Zooming is done with the Zoom toolbar buttons 

Click the  button. The button remains active until you press it again. The cursor changes into a magnifying glass. Use your mouse to select a sound segment. The selected segment will be expanded on screen so that it will fill the complete graphical window. Click on the same button again

to stop using the zoom-in cursor. The positioning tool above the graph shows the relative position of the currently zoomed part (in white) with respect to the complete file.



Click the  button. The button remains active until you press it again. The cursor changes into a magnifying glass. Use your mouse to select a screen section that will contain what currently fills the complete graphical window. Click the same button again to stop zooming out.

Click the  button and the complete sound file will be displayed in the graph window (no zoom factor).

5.1.7. Scrolling a zoomed graph

Scrolling is done with the Scroll toolbar buttons   

Click the  button. The current visible window will slide to the left in the complete file.

Click the  button. The current visible window will slide to the right in the complete file.



When you scroll left or right the positioning tool above the graph shows the relative position of the currently zoomed part (in white) with respect to the complete file.

Click the  button and the cursor will change into a hand. You can now manually drag the current visible window within the complete file. Click the same button again to stop scrolling manually.

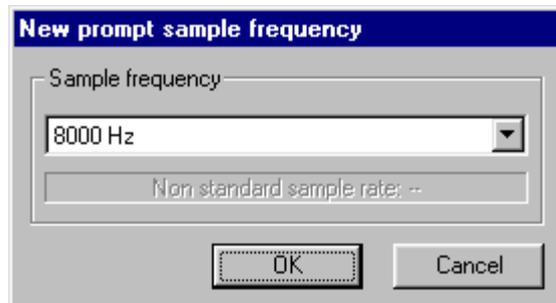
5.2. File menu

This menu is used to open, close or save the files you work with, to view file properties or to terminate the program session.

[New](#)
[Open](#)
[Close](#)
[Close All](#)
[Save](#)
[Save As](#)
[Save All](#)
[Properties](#)
[Exit](#)

5.2.1. New

This command creates a new untitled workspace to edit or record into. The first question that is asked is the sample frequency you want to use:



Select one of the standard frequencies suggested, or enter your own custom sample frequency. A new, but empty and untitled graph window is then created in the graph pane. You can now paste sound, generate DTMF or record sound into this area. The empty and unused sound window looks like this:



The New command can also be activated by clicking on  in the toolbar section.

5.2.2. Open

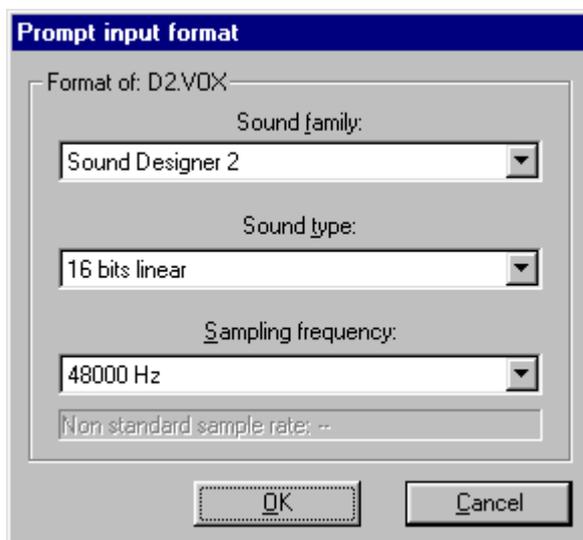
This command loads one or more files in Vox Studio. It opens a dialog box from which you can browse to and select the files you want to load.



By default Vox Studio looks for files in the directories defined in the Tools/Defaults/Folder defaults menu. Vox Studio also looks by default for ".wav" files or telephony files as defined in the same default menu. You can change these defaults at any time, or you can change the proposed default on-the-fly while using Vox Studio. This behavior is intended to simplify the work of those who have to manipulate thousands of files all located in the same folder and all of the same type. If you just want to edit a single file this behavior may not be that helpful.

If the file you open is a file type which has a recognizable header (".wav" and ".vsn" files have such headers for instance) then Vox Studio gets all the file information it needs from the header portion of the file and will not bother you with additional questions.

Vox Studio, however, cannot guess what kind of sound data you have in your files unless it is explicitly written in the file or unless you tell Vox Studio what it is. Therefore, if the file you open is a headerless file (Dialogic ".vox" for instance) or a file with a non-recognizable header, then, when needed, Vox Studio will open a dialog box and request additional information about the file such as sample rate and coding format.



Telephony format files often lack a header (luckily there are a few exceptions) and the program user is the only one able to provide the information that Vox Studio needs to read the file data correctly. It is, of course, crucial to give Vox Studio the right file information! If you do not, Vox Studio will still read

the file but what you will hear or see will be junk. Typically, if you hear screeching noise instead of normal sounds, then you have probably selected the wrong coding algorithm (ADPCM instead of A-law for example). If your file plays too fast or too slow, then you have given the wrong sample rate to Vox Studio.

The Open command can also be activated by clicking on  in the toolbar section.

5.2.3. Close

This command closes the active window. If the sound file in the window has been modified since it was last saved you will be prompted to save the file to disk.

The Close command can also be activated by clicking on  in the toolbar section.

5.2.4. Close All

This command closes all the loaded files. If the sound file in any window has been modified since it was last saved you will be prompted to save that file to disk.

The Close All command can also be activated by clicking on  in the toolbar section.

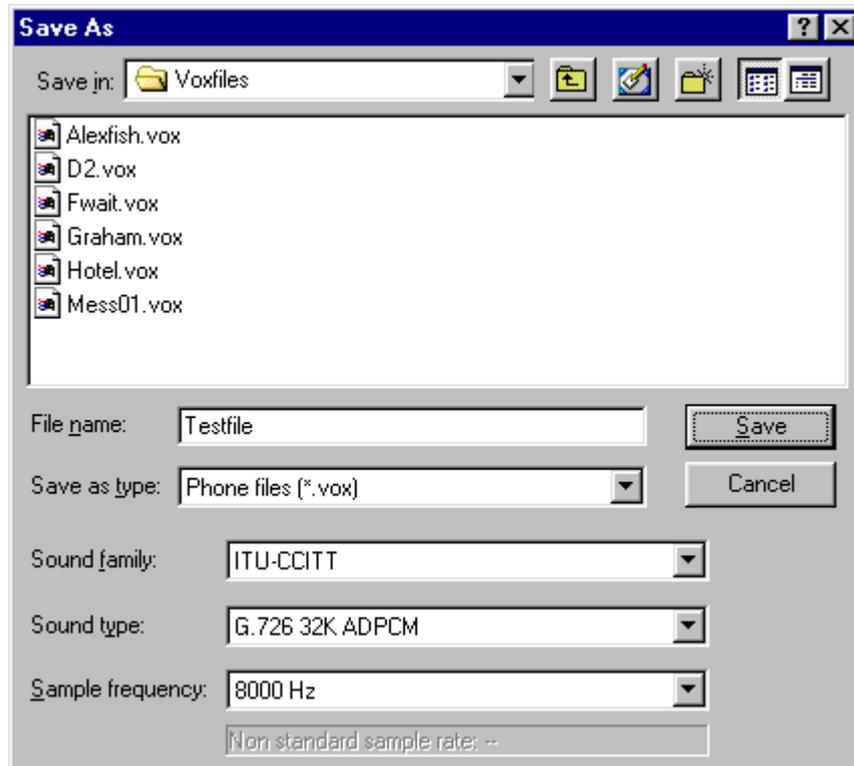
5.2.5. Save

This saves the file in the active window to its original location on disk. The file is saved under its original name and in its original sound format.

The Save command can also be activated by clicking on  in the toolbar section.

5.2.6. Save As

This opens a dialog box. You can browse to the location where you want to save the file and you can also select the filename and extension, the sound family, the sound type and the sample frequency to save the file under. If the selected file format is different from the original one a conversion is automatically done by Vox Studio.



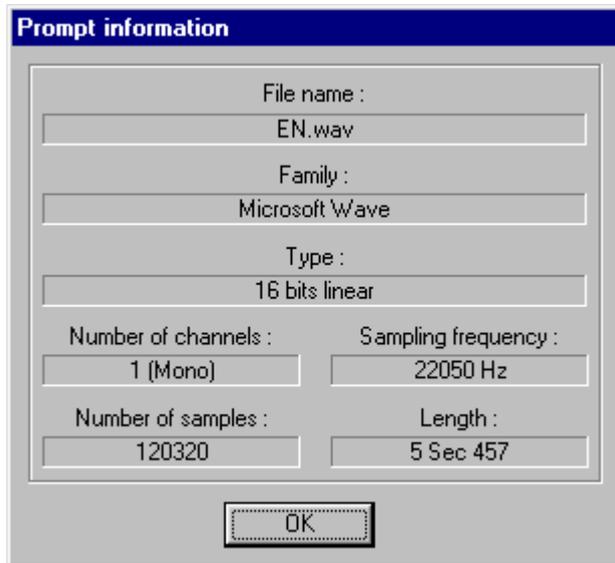
5.2.7. Save All

This saves all the files loaded in Vox Studio to their original location on disk. The files are saved under their original names and in their original sound formats. Naturally, only files that have been modified are saved.

The Save All command can also be activated by clicking on  in the toolbar section.

5.2.8. Properties

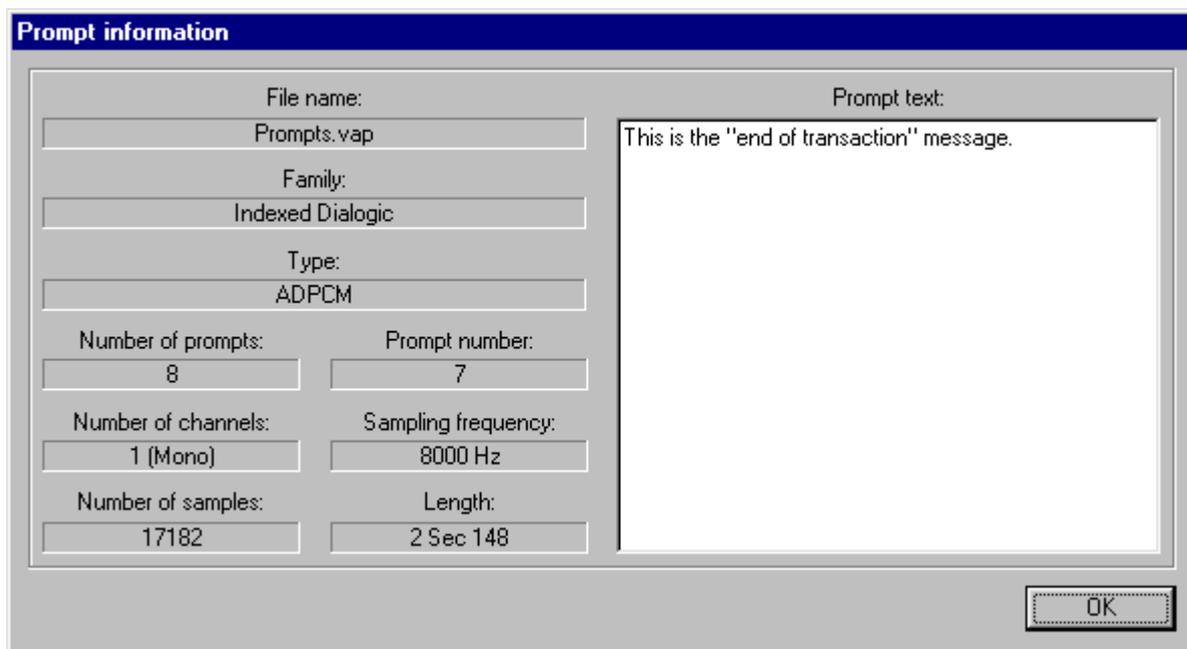
This opens an information window giving details about the active file:



The information provided is:

- File name
- Sound Family
- Sound Type
- Number of sound channels
- Sound sampling frequency (sample rate)
- Total number of sound samples
- Total sound length in seconds

The properties window is slightly different for indexed files. For instance, if the current indexed file format allows text comments within the indexed file, then the comment for the currently selected prompt will also be shown in the properties window. It may look like this:



The Properties command can also be activated by clicking on  in the toolbar section.

5.2.9. Exit

This command terminates Vox Studio and exits to Windows. If files need to be saved, you will be prompted to do so. You can terminate the program using these 6 normal methods:

You can use the menu command File/Exit.

You can click the green exit button on the toolbar

You can click the X button in the top right corner of Vox Studio.

You can click the tiny icon in the top left corner of Vox Studio and select close from the menu.

You can press Alt-F4 on your keyboard when working in Vox Studio.

You can right click on the program icon on the taskbar and select "Close" from the pop-up menu.

The Exit command can also be activated by clicking on  in the toolbar section.

Panic closure: The following method should only be used if none of the normal methods above work and when, for some reason, Vox Studio is frozen for several minutes and does not react to any command. This should normally never happen and you should contact us if it does.

Press and hold Ctrl-Alt-Del on your keyboard, release the keys. Select Vox Studio and click "End task". Vox Studio will terminate but all your unsaved work in Vox Studio and some changes to the defaults will be lost. Again this is for program crashes only. In NT you would have to start the task manager by right-clicking in the Task bar and then kill the process.

5.3. Edit menu

This menu is used to perform manual cut-and-paste operations on individual files. Most commands are also accessible via toolbar buttons.

[Undo](#)

[Cut](#)

[Copy](#)

[Paste](#)

[Delete](#)

[Select All](#)

5.3.1. Undo

This command undoes one or more previous edit commands. The number of undo levels available is settable in the Vox Studio program options using the Tools/Options/System command. The maximum number of undo levels is 8.

The Undo command can also be activated by clicking on  in the toolbar section. You can also use the Ctrl-Z keyboard shortcut to undo.

5.3.2. Cut

This command cuts (removes) a selected segment of sound from the active window. This only works if a segment of sound has been selected. The cut segment will be placed on the program's clipboard so that it can be pasted elsewhere in the same or in another sound file.

The Cut command can also be activated by clicking on  in the toolbar section. You can also use the Ctrl-X keyboard shortcut to cut.

5.3.3. Copy

This command copies (but does not remove) a selected segment of sound from the active window. This only works if a segment of sound has been selected. The copied segment will be placed on the program's clipboard so that it can be pasted elsewhere in the same or in another sound file.

The Copy command can also be activated by clicking on  in the toolbar section. You can also use the Ctrl-C keyboard shortcut to copy.

5.3.4. Paste

This command pastes a segment of sound from the program's clipboard into the active file at the current cursor position or into the current selection. The selected section will be replaced by the pasted data. If no cursor position or selection exists the segment of sound will be pasted at the end of the active file. The segment of sound remains on the program's clipboard and can be pasted at other places until it is replaced by another segment of sound or until the program is closed.

The Paste command can also be activated by clicking on  in the toolbar section. You can also use the Ctrl-V keyboard shortcut to paste.

5.3.5. Delete

This command deletes a selected segment of sound from the active window. If nothing is selected in the active window, nothing will be deleted. The deleted segment of sound is discarded and is not placed on the clipboard.

The Delete command can also be activated by clicking on  in the toolbar section.

5.3.6. Select All

This selects the complete sound file in the active window. You can then delete it, cut it or paste something else in its place.

5.4. View menu

This menu is used to select what part of a sound file is viewed on-screen and how much detail you want to see.

[Window Limits](#)

[Zoom In](#)

[Zoom Out](#)

[Zoom All](#)

[Scroll](#)

[Scroll Left](#)

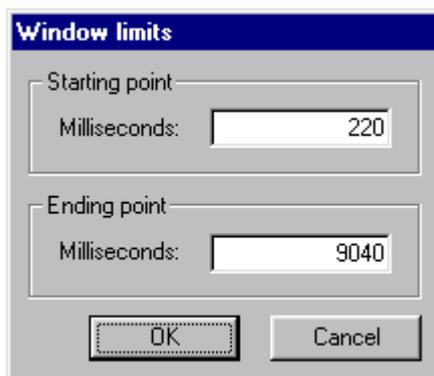
[Scroll Right](#)

[Previous View](#)

[Next View](#)

5.4.1. Window Limits

This command opens the following window:



The command is equivalent to the manual interactive Zoom In or Zoom Out commands but allows very precise entry of the beginning and end point of the zoomed-in waveform displayed on-screen. If the current graph display mode is in time units, then milliseconds will be shown for the starting point and ending point. If the current graph display mode is in sample units, then the first and last sample numbers will be shown.

5.4.2. Zoom In

A new cursor with a magnifying glass and a "+" sign appears. You can now interactively select the portion of the sound file you want to see on-screen. The zoom-in factor is defined like this:

The area selected by the magnifying glass cursor will be displayed in the entire graphical window.

The Zoom In command can also be activated by clicking on  in the toolbar section.

5.4.3. Zoom Out

A new cursor with a magnifying glass and a "-" sign appears. You can now interactively select the portion of the sound file you want to see on-screen. The zoom-out factor is defined like this:

The entire graphical window will be displayed in the area selected by the magnifying glass cursor.

The Zoom Out command can also be activated by clicking on  in the toolbar section.

5.4.4. Zoom All

This restores the display so that it shows the complete waveform in the graphical window (no zooming).

The Zoom All command can also be activated by clicking on  in the toolbar section.

5.4.5. Scroll

This only makes sense if the file has been zoomed. A new cursor with a hand appears. You can now drag left or right to interactively slide to the portion of the sound file you want to see on-screen. A position bar at the top of the graphical window shows the relative position of the display with respect to the complete file and also shows the current selection, if any exists.

The Scroll command can also be activated by clicking on  in the toolbar section.

5.4.6. Scroll Left

This only makes sense if the file has been zoomed. You can now click to interactively move left to the portion of the sound file you want to see on-screen. Each click moves the position by 80% of the total visible window. A position bar at the top of the graphical window shows the relative position of the display with respect to the complete file and also shows the current selection, if any exists.

The Scroll Left command can also be activated by clicking on  in the toolbar section.

5.4.7. Scroll Right

This only makes sense if the file has been zoomed. You can now click to interactively move right to the portion of the sound file you want to see on-screen. Each click moves the position by 80% of the total visible window. A position bar at the top of the graphical window shows the relative position of the display with respect to the complete file and also shows the current selection, if any exists.

The Scroll Right command can also be activated by clicking on  in the toolbar section.

5.4.8. Previous View

This command simply shows the previous graphical screen(s), so that you can go back to a previous zoom or scroll position.

This command loops. If you click one more time after the first view you will see the last view.

The Previous View command can also be activated by clicking on  in the toolbar section.

5.4.9. Next View

This command simply shows the next graphical screen(s), so that you can go back to another zoom or scroll position. This only works if one or more "Previous Screen" commands has been issued before.

This command goes full cycle. If you click one more time after the last view you will see the first view.

The Next View command can also be activated by clicking on  in the toolbar section.

5.5. Prompters menu

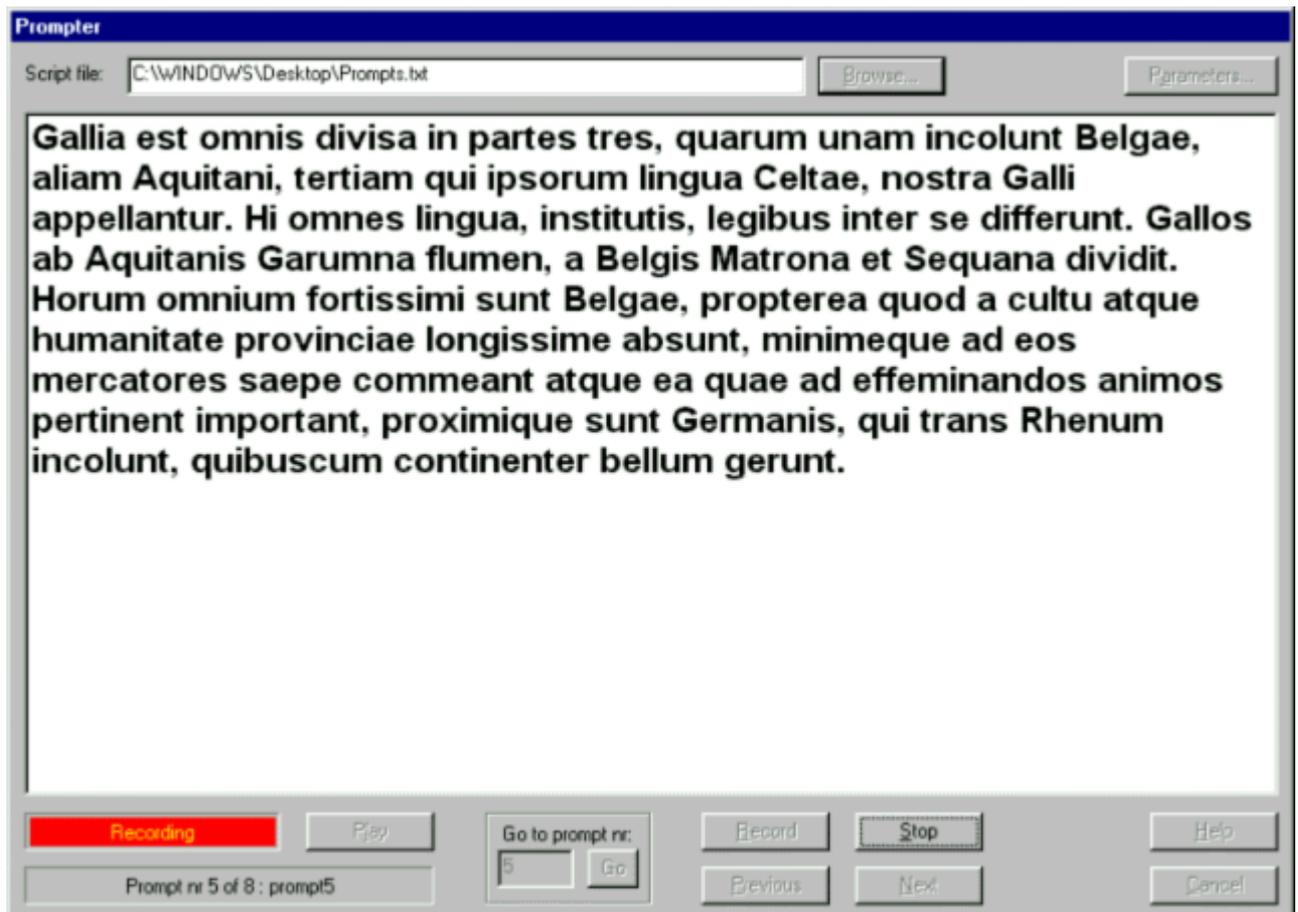
The prompter and the tape loader are somewhat similar. Both allow very fast recording of a large

number of messages. They work from microphone or from a tape input. The file slicer allows slicing of a large ".wav" file containing lots of concatenated prompts.

[Prompter](#)
[Tape Loader](#)
[File Slicer](#)

5.5.1. Prompter

The Prompter command is used to very rapidly record a large number of prompts.



The prompter works with a prompt script file. The script file contains the prompt text, the names 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. Prompts can be rerecorded easily.

The prompter flashes the script's messages on-screen in a large typeface. This enables the speaker (voice talent) to read them, one by one, into the microphone and save them automatically under the file names previously defined in the script file. This is a real time-saver.

Recorded messages can be played back immediately for verification from within the prompter, before proceeding to the next prompt.

The script file format is described in detail elsewhere. 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 recorded prompts.

You can change the default display font for the Prompter by setting it in the Tools/Options/Prompter font dialog box.

The Prompter command can also be activated by clicking on  in the toolbar section.

[Prompter options](#)

5.5.2. Prompter options

This is the dialog box where you define which type of ".wav" file the prompter will produce. You can choose between 8 and 16-bit files and sample rates from 6 to 48 KHz. If you don't know where to start select 16-bit files at 22 KHz, and go from there.

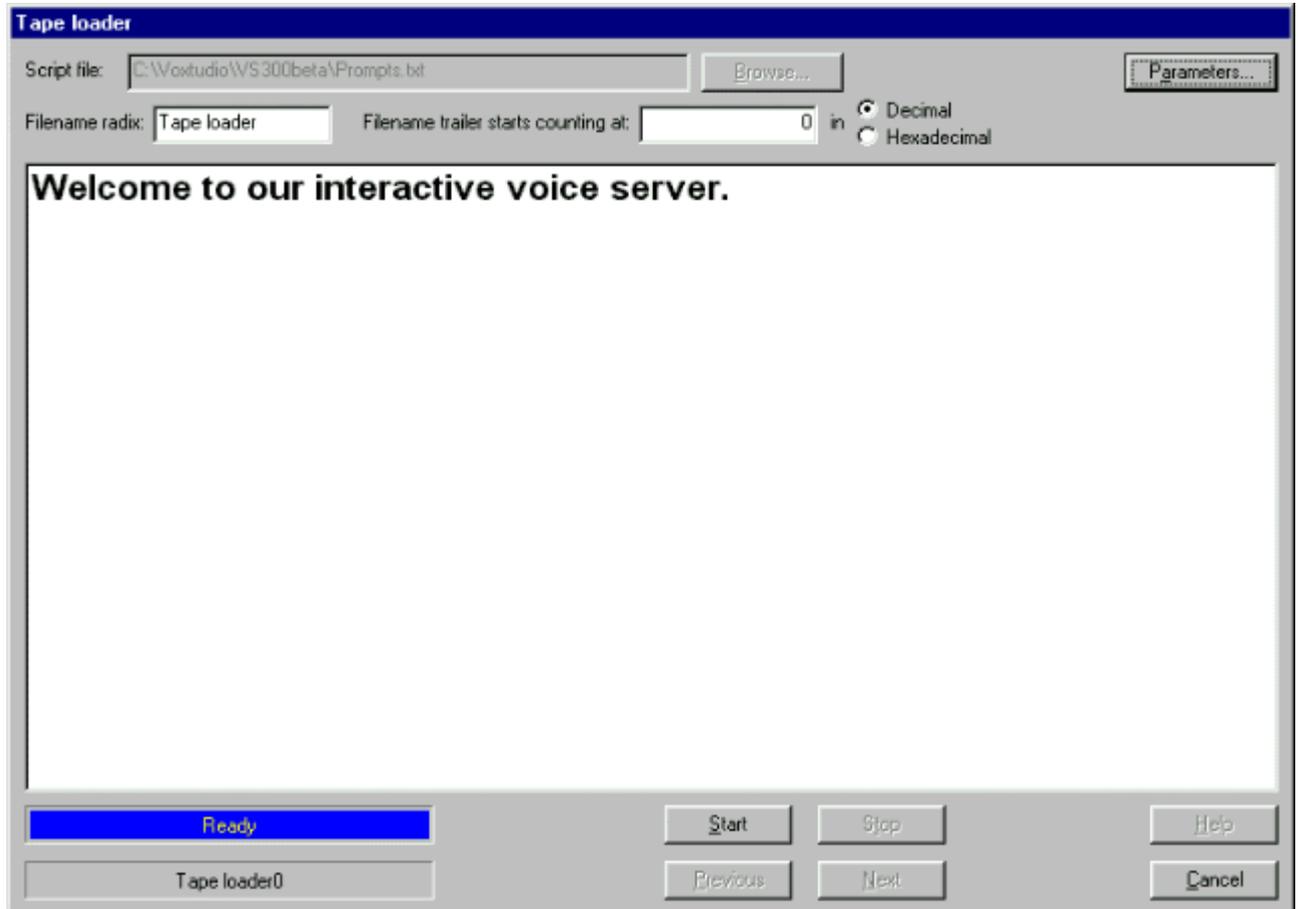
Browsing buttons allow you to select the script file that will be used for the next prompter session as well as the target directory where the recorded prompts will be saved. If you don't enter a "save-to" path in the prompter defaults then the complete path with file name (without extension though, it is always ".wav") has to be entered in the script file itself. If a save-to directory is entered in the prompter defaults only a file name (without extension) should be entered in the script file.



The prompter parameters can also be set as system defaults through the Tools/Defaults menu so that next time you use the prompter the same selections will be presented as defaults.

5.5.3. Tape Loader

The tape loader is similar to the Prompter but is specifically intended for the digitization of prerecorded studio tapes or CDs.



The tape loader tool can automatically digitize a prerecorded "studio" tape or CD. Connect the output of your tape or CD player to the line input of your sound card. The tape loader will automatically detect the silent delimiting spaces between recorded prompts, cut your recordings there and save the digitized data in preassigned filenames, as defined by an ASCII script file. Again, this script file is fully compatible with the prompter and the group/ungroup script files. The loader can also automatically create incrementing filenames, which relieves you from the task of creating a script file. Once correctly set up and started, the tape loader does all the digitizing without operator intervention.

While the tape loader digitizes your tape or CD it shows you on-screen what it is doing, so that, once in a while, you can reassure yourself that all is going well. For the tape loader to work well your recordings need to be clean, without any background noise in the silent passages. It is also crucial that you very carefully select the threshold level at which Vox Studio separates silence from non-silence. This is a settable parameter. If that accuracy is not achievable, you can still use the tape loader in its semiautomatic mode where you simply tap the space bar when you hear the delimiting silence between two prompts. If you have background noise in your original recordings or if you have clean recordings but fail to correctly select the silence detection threshold, you may find it very hard to obtain consistent results with the tape loader in automatic mode.

When used with a direct microphone connection rather than connected to the output of a tape or CD

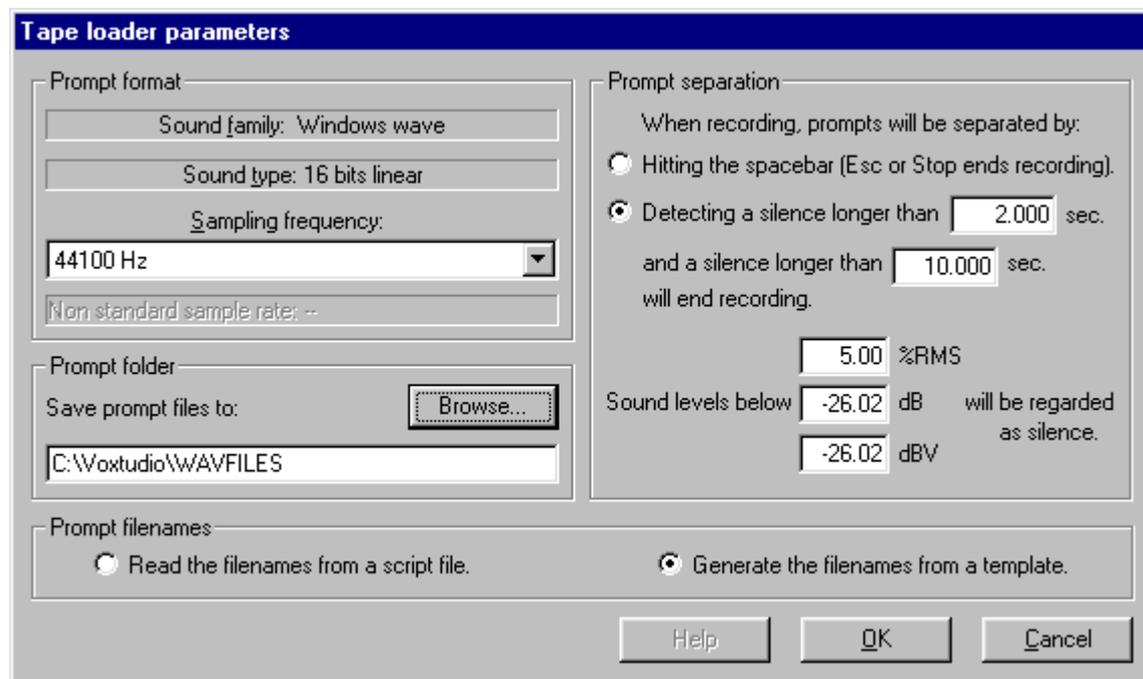
player you can even use the tape loader as a sort of voice-operated prompter. Speak a message, remain silent, speak the next message, and so on. To facilitate the use of the tape loader as a voice-operated prompter the tape loader will show you the next script text to be recorded on-screen as soon as it detects the end of the previous one. This gives you enough time to read the first sentence mentally before you start speaking it. This is the fastest way to record prompts in real time.

The Tape Loader command can also be activated by clicking on  in the toolbar section.

[Tape loader options](#)

5.5.4. Tape Loader options

This is where you set the operating parameters for the Tape Loader



Select the filenames under which the recorded messages will be saved. You can browse to an existing script file, in which case the filenames are taken from the script file itself. If you don't enter a "save-to" path in the tape loader defaults then the complete path (without extension though; it is always ".wav") has to be entered in the script file itself. If a save-to directory is entered in the loader defaults only a filename (without extension) should be entered in the script file.

Alternatively you can have Vox Studio generate the filenames automatically. The computed filenames will consist of a fixed radix (root) and a variable (incremented) digital tail (suffix). The counting can be decimal or hexadecimal, as preferred. If you don't know what hexadecimal notation is, just select decimal. For example if the starting filename template's radix is ivrsys and the trailing counter is 00, then filenames from ivrsys00.wav to ivrsys99.wav can be generated. If the starting radix is ivr and the trailing number is 1087, then filenames from ivr1087.wav to ivr9999.wav will be generated. If you have selected to automatically generate filenames, then the default filename prefix and suffix will be proposed at the top of the window and you can modify it there if you want.

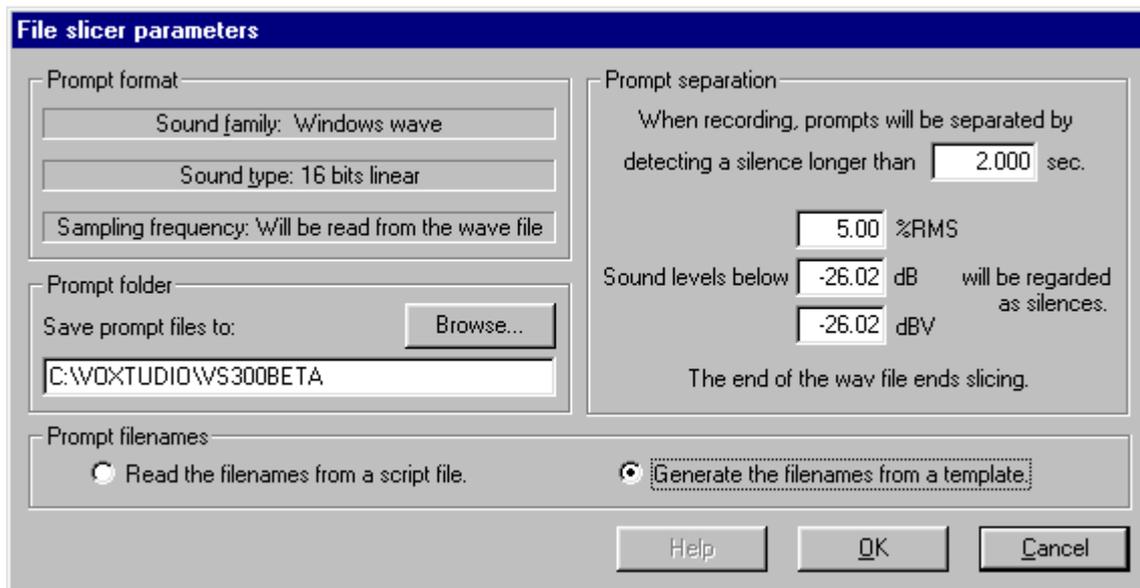
The next step is to decide how the tape loader is going to detect the separation between recorded

messages. You can make it simple for Vox Studio and tap the space bar yourself whenever you recognize the end of a prompt. Alternatively, you can have Vox Studio do this for you by detecting a predefined (and settable) number of seconds of silence. It would be a good idea to have 3 to 5 seconds of silence as a delimiter on your tape recordings. Of course, you will have to direct the recording studio to do that for you. You can also select what length of silence represents the end of all tape recordings. This should be much longer than the longest separation on the tape. Finally, you can also set the threshold level Vox Studio is going to use to make the difference between silence and speech (or speech and silence). You can use a setting in %RMS or in decibels.

Finally, select the sample rate of the recorded ".wav" files (6 to 48 KHz) and the disk directory where these will be stored.

5.5.5. File Slicer

This tool is used to cut one large prerecorded ".wav" file into smaller files. Silences are used as prompt delimiters.



Select the filenames under which the recorded messages will be saved. You can browse to an existing script file, in which case the filenames are taken from the script file itself. If you don't enter a "save-to" path in the tape loader defaults then the complete path (without extension though, it is always ".wav") has to be entered in the script file itself. If a save-to directory is entered in the loader defaults only a filename (without extension) must be entered in the script file.

Alternatively you can have Vox Studio generate the filenames automatically. The computed filenames will consist of a fixed radix (root) and a variable (incremented) digital tail (suffix). The counting can be decimal or hexadecimal, as preferred. If you don't know what hexadecimal notation is, just select decimal. For example, if the starting filename template's radix is ivrsys and the trailing counter is 00, then filenames from ivrsys00.wav to ivrsys99.wav can be generated. If the starting radix is ivr and the trailing number is 1087, then filenames from ivr1087.wav to ivr9999.wav will be generated. If you have selected to automatically generate filenames, then the default filename prefix and suffix will be proposed at the top of the window and you can modify it there if you want.

The next step is to decide how the file slicer is going to detect the separation between recorded

messages. You can have Vox Studio do this for you by detecting a predefined (and settable) number of seconds of silence. It would be a good idea to have 3 to 5 seconds of silence as a delimiter on your recordings. Of course, you will have to direct the recording studio to do that for you. You can also select what length of silence represents the end of all tape recordings. This should be much longer than the longest separation on the tape. Finally, you can also set the threshold level Vox Studio is going to use to detect the difference between silence and speech (or speech and silence). You can use a setting in %RMS or in decibels.

Finally, select the disk directory where the new files will be stored.

The File Slicer command can also be activated by clicking on  in the toolbar section.

5.6. Convert menu

The conversion tools are all found here:

- Convert Active allows in-memory manual conversion of a currently-loaded and active file.
- Convert Current allows manual conversion of a currently-loaded file, and it also saves the file to disk.
- Batch Conversion, as the name implies, allows automated conversion of a whole bunch of files.
- The remaining commands allow concatenation of single prompts to grouped (indexed) files or extraction of indexed files into their separate components. Grouped (indexed) files contain many prompts and are very useful because the operating system under which the voice application runs does not have to keep hundreds of files open at any time. For instance, it would make sense to keep recordings for all the numbers from 1 to 99 into one and the same indexed file. Similarly, all error messages could be kept in one larger indexed file too.

[Transform Active](#)

[Convert Active](#)

[Batch Convert](#)

[Group to indexed Dialogic](#)

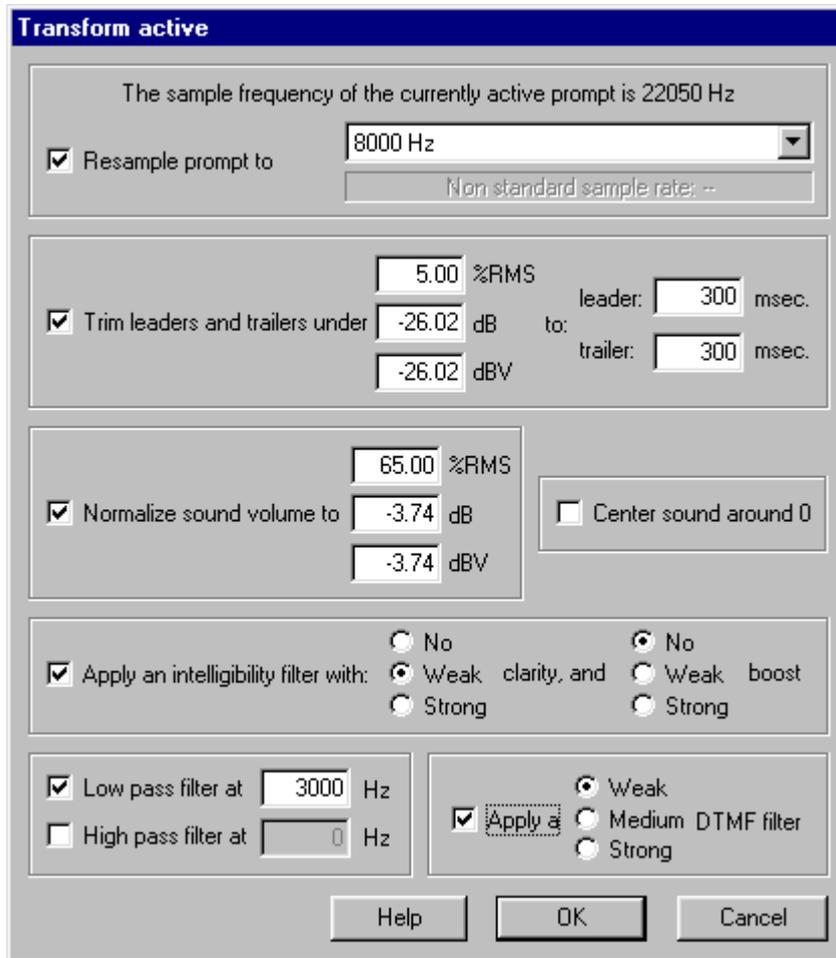
[Ungroup from indexed Dialogic](#)

[Group to indexed NMS](#)

[Ungroup from indexed NMS](#)

5.6.1. Transform Active

The transform active command modifies the active file currently in memory. It does not alter the original file on disk until you actually save the modified data. Transform active can be tested in real time by playing the file after modification. Transform active operations can be undone with the Edit/Undo command.



All the default values to this dialog box can be preset using the Tools/Defaults menu.

- Resample prompt:** modifies the sample rate of the file, i.e. the number of signal measurements performed per second. According to the Nyquist theorem the highest frequency component in any sound file cannot exceed 1/2 the frequency at which the signal is sampled. So, when a file is down-sampled (re-sampled at a lower sampling rate) Vox Studio automatically filters the file to remove the components above 1/2 the sampling frequency before doing the re-sampling. Naturally, when a file is up-sampled (re-sampled at a higher sampling rate), there is no way Vox Studio can add frequency components that are not present in the file (or have been removed by previous down-sampling). In other words, down-sampling will necessarily result in a file with a lower bandwidth, but up-sampling will not increase the file's frequency content. Thus, when you do a down-sampling followed by a reverse up-sampling the resulting file will not be the original. If that is the intent, you should reload the original file from disk instead. Downsampling irretrievably deteriorates a file and should always be the last thing you do to a file. You can select any of the "standard" telephony or multimedia sample rates from a list, but you can also input a custom "non-standard" sample rate if you need to.
- Trim leaders and trailers:** allows easy production of prompt files with uniform leading and trailing blanks. What used to be a time-consuming manual editing task has now become as easy as selecting a button with your mouse. The rest is done for you automatically.

Automatic silence adjustment is a threshold-activated process and thus requires spotless, clean recordings. If you have background noise in your recordings, Vox Studio may incorrectly detect the beginning and end of sound in your file.

The dialog box allows you to select the threshold level Vox Studio uses to detect the difference between silence and non-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 based on the threshold you selected. 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 threshold level can be set either as a percentage of the maximum on-screen amplitude or as a threshold value in dB, dBm or dBV.

The silence duration itself is defined in the Tools/Defaults/Convert Active menu. A reasonable typical value would be 300 milliseconds. Note that the Leader/Trailer option is actually capable of adding silence to your file if you request more leading or trailing silence than your file actually contains!

If the voice files you are trimming are not recorded in optimal conditions it may be very useful to perform a "Center" or a "Normalize" operation on your file before you apply the "Leader/Trailer" command. This has the effect of "flattening" the low frequency or DC signal background on the zero line and "pumping up the signal" which makes it so much easier to perform good threshold detection on your signal.

- **Normalize sound volume:** allows easy and rapid production of prompt files with equal (and, for most telephony applications, preferably reasonably high) sound volume levels.

The dialog box allows you to select the maximum sound energy desired. This volume selection can be done in % of the maximum on-screen amplitude, in dB or in dBm/dBV. The current file will be scanned for maximum sound energy levels and the whole file will be multiplied by a factor that will bring the maximum sound energy to the desired level. Vox Studio does not measure peak AMPLITUDE, it measures peak ENERGY over a duration of several milliseconds (about the duration of a spoken syllable). Vox Studio levels are internally calibrated for steady pure sine waves at 800Hz. This may confuse you in the beginning as the best setting in Vox Studio usually is a choice of about 65% of maximum energy, not 100%. That's simply because human voice is neither steady nor a pure sine wave. On a pure steady sine wave a Vox Studio normalization to 100% will produce a signal that just reaches the floor and the ceiling of the display. As a result, if you have recorded speech and you select a maximum average level of energy of say 70% in the Normalize command you can get sound spikes whose amplitude will very briefly be clipped at 100%. For voice recordings, this happens on very short sounds like "T", "P" and "K" which are explosive sounds anyway and where amplitude saturation does not matter very much. This technique allows recording at as high a level as possible, with best possible telephony quality. A similar technique is used when recording with a recording device that has a recording meter and adjusting the level to remain below 0 dB most of the time (green area lights) but allowing some very short spikes to exceed 0 dB (red area lights briefly).

The best practical approach is to calibrate your Vox Studio recording and playback setup once and for all by fine-tuning your recording settings. You can use the monitor function and the graphical display function to do that. Your signal should go from time to time in the red range on the Monitor VU-meter and should use at least 75% of the available amplitude range shown on the graphical waveform display screen. It does not matter too much if your signal peaks sometimes and briefly (we mean briefly) hits the ceiling or the floor. This usually will happen for utterances with T, P and K where a little clipping does not matter so much. You should select a preferred normalization factor that gives best volume and quality results when played back to the specific telephony card you use. Once that is set, your settings should usually never change unless you change the target telephony system. Remember that the Normalize option is there to correct minor variations between recordings; it is not there to correct grossly clipped or grossly under-recorded signals. You should regularly check if you use nearly the full dynamic range you have available by looking at your recorded messages in the graphical display window. Note: when Normalize is selected, a Center function is in fact performed automatically on the signal prior to amplitude normalization.

Finally, although this functionality is described elsewhere in the chapter describing the calibration

tool, remember that Vox Studio's dBV or dBm display capability can be calibrated so that the dB levels you see on-screen actually match the dB levels that the same signal will generate on the target telephone line.

- **Center around zero:** automatically re-centers the sound file around the zero baseline to eliminate DC offsets caused by your sound card, and possibly by very-low-frequency interference. Note: when a Normalize command is issued, a Center command is, in fact, performed automatically on the file prior to amplitude normalization.

Depending on the quality of your sound card, and your recording setup, it may also be very advantageous to perform a centering operation before doing leader/trailer work (which requires rather precise threshold detection). Vox Studio does not automatically perform a centering operation before every leader/trailer command but you may select this option.

You can perform a quick visual check using the graphical waveform display in the main window. Record a file with nothing but silence using your onboard sound card. Save the file and then reload it into Vox Studio. You should see a clean horizontal line exactly superimposed on the zero baseline. If you see two lines your hardware does generate significant offset. If you see only one line the offset generated is small enough that you can disregard it. If you see green rubbish (grass) on the base line your recording setup picks up unwanted noise (which you will certainly hear in the final recording).

If you manipulate files that were recorded on another workstation, it is a good idea to load the files first and visually check them for a possible DC shift.

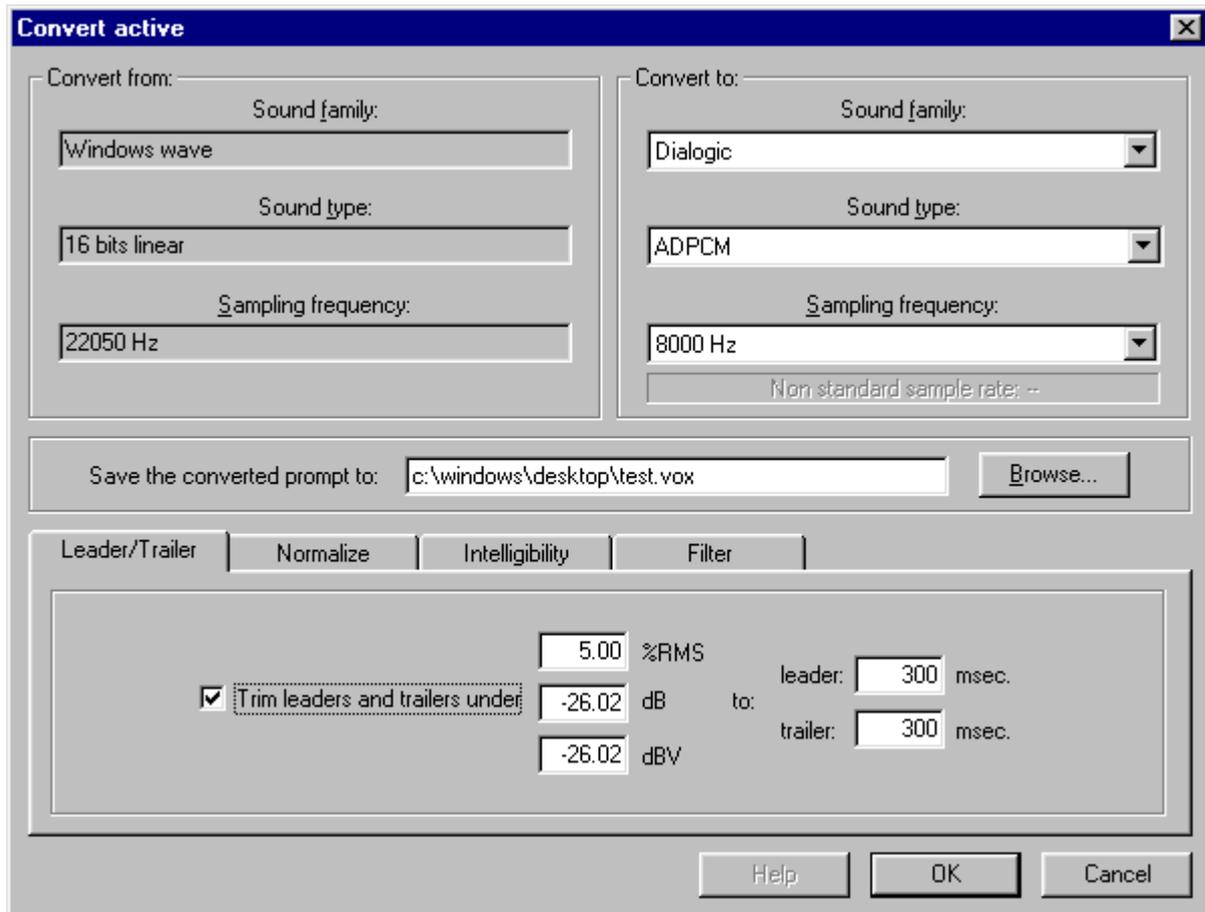
DC offsets are very disturbing whenever you need to amplify a signal or do threshold detection. Also too much DC offset can generate audible clicks at the beginning and end of a file.

- **Intelligibility filter:** consists of a Clarity filter which corrects the muffled sound effect often obtained when down-sampling voice files and a Boost option which produces variable signal amplification in order to increase the perceived voice energy content of the recorded file. The Intelligibility Filter can be selected only while doing a Convert Active, Convert Current or a Batch Conversion operation. Intelligibility filter options are set in the defaults section for the conversion functions. Usually a "weak" clarity filter and "no boost" is the best choice. Use trial-and-error to select the options that suit your voice files best.
- **Low- and high-pass filter:** allows individual selection of either a high-pass (low-cut) or a low-pass (high-cut) filter. For the high-pass and low-pass filters the -3 dB cutoff frequencies can be chosen with a 1 Hz resolution. According to the Nyquist theorem the highest frequency component in any sampled file cannot exceed 1/2 the frequency at which the file is sampled. Audible "aliasing" errors occur whenever the Nyquist theorem is overlooked. It makes little sense to select a cutoff frequency above 1/2 the sampling frequency of the file.
- **DTMF filter:** removes signal components at and near the frequencies that correspond to one group of the DTMF frequencies. This harsh treatment effectively removes talk-off problems, but also degrades the sound quality somewhat. Use this only on files that really do cause talk-off problems and cannot be rerecorded. The DTMF filter has 3 strengths you can choose from: weak attenuates 10 dB, medium attenuates 20 dB, and strong attenuates 30 dB. Use the weakest filter that solves your talk-off problem.

The Transform Active command can also be activated by clicking on  in the toolbar section.

5.6.2. Convert Active

This command is similar to the previous Transform Active command except that it immediately saves a converted file to disk. You can convert from any format known to Vox Studio to any other format known to Vox Studio. You can do up-sampling conversions and down-sampling conversions. See [File Formats](#) for an enumeration of all the formats Vox Studio currently handles. You can do file conversions in Vox Studio without any sound card installed on your system. Of course, you will need a sound card to record or play the files on your PC, before or after conversion.



All the default values to this dialog box can be preset using the Tools/Defaults menu.

- **Convert from:** there is nothing to select here; the conversion source file is the currently loaded and active voice file.
- **Convert to:** you always have to fully describe the sound family and sound type you want to convert to; Vox Studio cannot guess this. If you don't feel like entering this every time you can preset this as a default using the Tools/Defaults menu. You do not need to describe the format from which you want to convert as, by definition, the current file already has been loaded into Vox Studio and therefore the program will already have retrieved the information it needs at file loading time.
- The sampling frequency can modify the sample rate of the file, i.e. the number of signal measurements performed per second. According to the Nyquist theorem the highest frequency component in any sound file cannot exceed 1/2 the frequency at which the file is sampled. So, when a file is down-sampled (resampled at a lower sampling rate) Vox Studio automatically filters the file to remove the components above 1/2 the sampling frequency before doing the resampling.

Naturally, when a file is up-sampled (resampled at a higher sampling rate), there is no way Vox Studio can add frequency components that are not present in the file (or have been removed by previous down-sampling). In other words, down-sampling will necessarily result in a file with a lower bandwidth, but up-sampling will not increase the file's frequency content. Thus, when you do a down-sampling followed by a reverse up-sampling the resulting file will not be the original. If that is the intent, you should reload the original file from disk instead. Downsampling irremediably deteriorates a file and should always be the last thing you do to a file. You can select any of the "standard" telephony or multimedia sample rates from a list, but you can also input a custom "non-standard" sample rate if you need to.

While converting you can also adjust leading and trailing silence, improve intelligibility, normalize the sound amplitude, or filter the sound. These options can be selected using the tabbed section of the dialog box.

- **Trim leaders and trailers:** allows easy production of prompt files with uniform leading and trailing blanks. What used to be a time-consuming manual editing task has now become as easy as selecting a button with your mouse. The rest is done for you automatically.

Automatic silence adjustment is a threshold-activated process and thus requires spotless, clean recordings. If you have background noise in your recordings, Vox Studio may incorrectly detect the beginning and end of sound in your file.

The dialog box allows you to select the threshold level Vox Studio uses to detect the difference between silence and non-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 based on the threshold you selected. 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 threshold level can be set either as a percentage of the maximum on-screen amplitude or as a threshold value in dB, dBm or dBV.

The silence duration itself is defined in the Tools/Defaults/Convert Active menu. A reasonable typical value would be 300 milliseconds. Note that the Leader/Trailer option is actually capable of adding silence to your file if you request more leading or trailing silence than your file actually contains!

If the voice files you are trimming are not recorded in optimal conditions it may be very useful to perform a "Center" or a "Normalize" operation on your file before you apply the "Leader/Trailer" command. This has the effect of "flattening" the low frequency or DC signal background on the zero line and "pumping up the signal" which makes it so much easier to perform a good threshold detection on your file.

- **Normalize sound volume:** allows easy and rapid production of prompt files with equal (and, for most telephony applications, preferably reasonably high) sound volume levels.

The dialog box allows you to select the maximum sound energy desired. This volume selection can be done in % of the maximum on-screen amplitude, in dB or in dBm/dBV. The current file will be scanned for maximum sound energy levels and the whole file will be multiplied by a factor that will bring the maximum sound energy to the desired level. Vox Studio does not measure peak AMPLITUDE, it measures peak ENERGY over a duration of several milliseconds (about the duration of a spoken syllable). Vox Studio levels are internally calibrated for steady pure sine waves at 800Hz. This may confuse you in the beginning as the best setting in Vox Studio usually is a choice of about 65% of maximum energy, not 100%. That's simply because human voice is neither steady nor a pure sine wave. On a pure steady sine wave a Vox Studio normalization to 100% will produce a signal that just reaches the floor and the ceiling of the display. As a result, if you have recorded speech and you select a maximum average level of energy of say 70% in the Normalize command you can get sound spikes whose amplitude will very briefly be clipped at 100%. For voice recordings, this happens on very short sounds like "T", "P" and "K" which are explosive sounds anyway and where amplitude saturation does not matter very much. This technique allows recording

at as high a level as possible, with best possible telephony quality. A similar technique is used when recording with a recording device that has a recording meter and adjusting the level to remain below 0 dB most of the time (green area lights) but allowing some very short spikes to exceed 0 dB (red area lights briefly).

The best practical approach is to calibrate your Vox Studio recording and playback setup once and for all by fine-tuning your recording settings. You can use the monitor function and the graphical display function to do that. Your signal should go from time to time in the red range on the Monitor VU-meter and should use at least 75% of the available amplitude range shown on the graphical waveform display screen. It does not matter too much if your signal peaks sometimes and briefly (we mean briefly) hits the ceiling or the floor. This usually will happen for utterances with T, P and K where a little clipping does not matter so much. You should select a preferred normalization factor that gives best volume and quality results when played back to the specific telephony card you use. Once that is set, your settings should usually never change unless you change the target telephony system. Remember that the Normalize option is there to correct minor variations between recordings; it is not there to correct grossly clipped or grossly under-recorded signals. You should regularly check if you use nearly the full dynamic range you have available by looking at your recorded messages in the graphical display window. Note: when Normalize is selected, a Center function is in fact performed automatically on the signal prior to amplitude normalization.

Finally, although this functionality is described elsewhere in the chapter describing the calibration tool, remember that Vox Studio's dBV or dBm display can be calibrated so that the dB levels you see on-screen actually match the dB levels that the same signal will generate on the target telephone line.

- **Center around zero:** automatically re-centers the sound file around the zero baseline to eliminate DC offsets caused by your sound card, and possibly by very-low-frequency interference. Note: when a Normalize command is issued, a Center command is, in fact, performed automatically on the file prior to amplitude normalization.

Depending on the quality of your sound card, and your recording setup, it may also be very advantageous to perform a centering operation before doing leader/trailer work (which requires rather precise threshold detection). Vox Studio does not automatically perform a centering operation before every leader/trailer but you may select this option.

You can perform a quick visual check using the graphical waveform display in the main window. Record a file with nothing but silence using your onboard sound card. Save the file and then reload it into Vox Studio. You should see a clean horizontal line exactly superimposed on the zero baseline. If you see two lines your hardware does generate significant offset. If you see only one line the offset generated is small enough that you can disregard it. If you see green rubbish (grass) on the base line your recording setup picks up unwanted noise (which you will certainly hear in the final recording).

If you manipulate files that were recorded on another workstation, it is a good idea to load the files first and visually check them for a possible DC shift.

DC offsets are very disturbing whenever you need to amplify a signal or do threshold detection. Also too much DC offset can generate audible clicks at the beginning and end of a file.

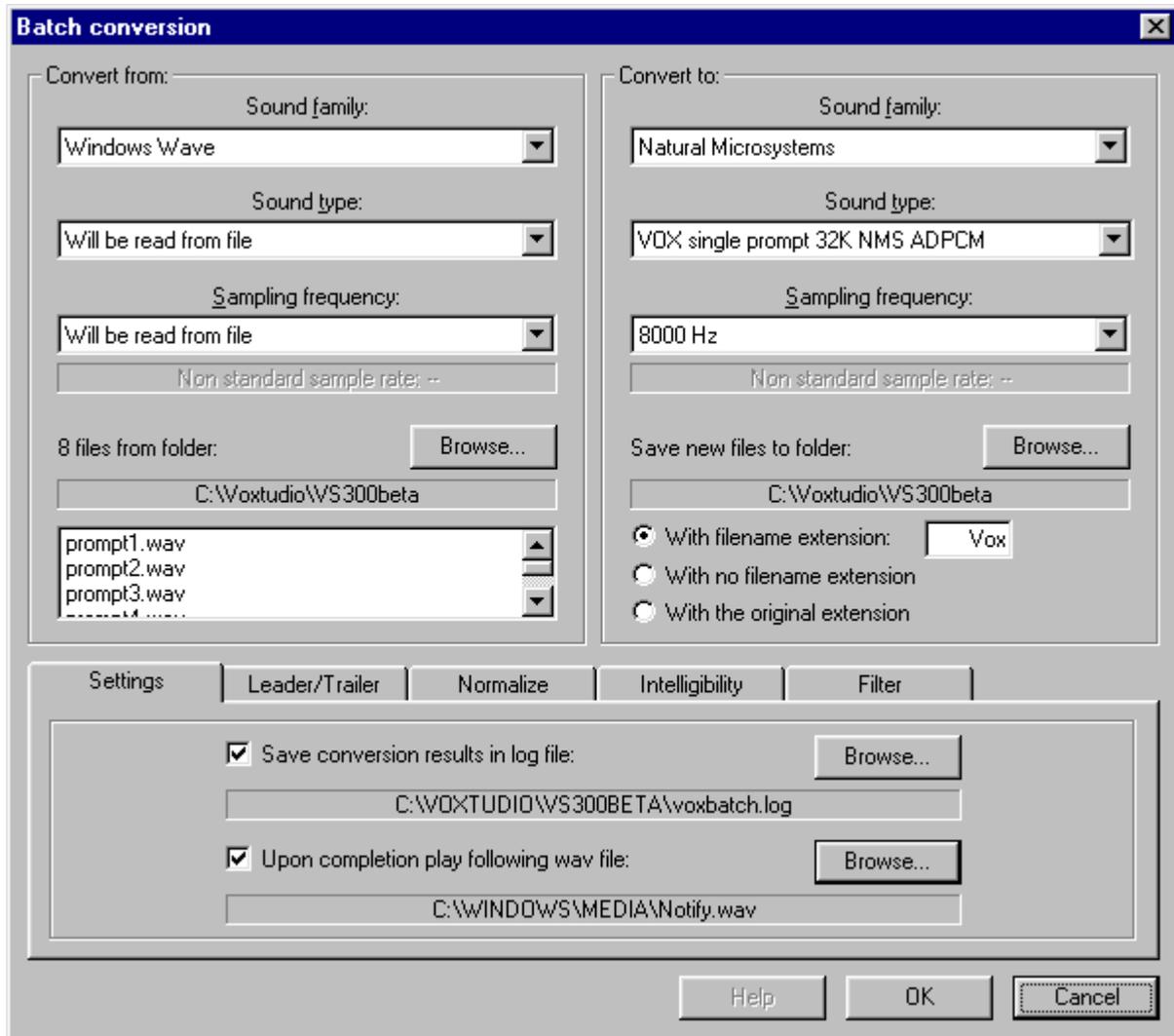
- **Intelligibility filter:** consists of a Clarity filter which corrects the muffled sound effect often obtained when down-sampling voice files and a Boost option which produces variable signal amplification in order to increase the perceived voice energy content of the recorded file. The Intelligibility filter can be selected only while doing a Convert Active, Convert Current or a Batch Conversion operation. Intelligibility filter options are set in the defaults section for the conversion functions. Usually a "weak" clarity filter and "no boost" is the best choice. Use trial-and-error to select the options that suit your voice files best.

- **Low- and high-pass filter:** allows individual selection of either a high-pass (low-cut) or a low-pass (high-cut) filter. For the high-pass and low-pass filters the -3 dB cutoff frequencies can be chosen with a 1 Hz resolution. According to the Nyquist theorem the highest frequency component in any sampled file cannot exceed 1/2 the frequency at which the file is sampled. Audible "aliasing" errors occur whenever the Nyquist theorem is overlooked. It makes little sense to select a cutoff frequency above 1/2 the sampling frequency of the file.
- **DTMF filter:** removes signal components at and near the frequencies which correspond to one group of the DTMF frequencies. This harsh treatment effectively removes talk-off problems, but also degrades the sound quality somewhat. Use this only on files that really do cause talk-off problems and cannot be rerecorded. The DTMF filter has 3 strengths you can choose from: weak attenuates 10 dB, medium attenuates 20 dB, and strong attenuates 30 dB. Use the weakest filter that solves your talk-off problem.

The Convert Active command can also be activated by clicking on  in the toolbar section.

5.6.3. Batch Convert

This command is similar to the previous Convert Active command except that it converts a whole bunch of files in batch mode and immediately saves the converted files to disk. You can convert thousands of files in one go. You can convert from any format known to Vox Studio to any other format known to Vox Studio. You can do up-sampling conversions and down-sampling conversions. See [File Formats](#) for a list of all the formats Vox Studio currently handles. You can do file conversions in Vox Studio without any sound card installed on your system. Of course, you will need a sound card to record or play the files on your PC, before or after conversion.



All the default values to this dialog box can be preset using the Tools/Defaults menu.

- **Convert from:** allows you to select the convert from file family, type and sample rate. If this information is written in the files themselves (in the file header) then Vox Studio will find the information automatically (like in ".wav" files). Many other file formats (especially the telephony formats) do not contain that information and then this has to be supplied to Vox Studio because the program has no other way of finding out what the input is. All input source files should have the same format. All converted output files will have the same format.

The browse button allows you to navigate to and select multiple files (literally thousands of files) or even a complete directory to convert. You can use the standard Windows file selection techniques to indicate which files need converting. Click left, Shift-Click left, CTRL-Click left all work as usual. See your Windows manual on how to select multiple files in a selection box. The good old *.* command from ancient DOS times works here too and selects all the files in the folder.

- **Convert to:** you always have to fully describe the sound family and sound type you want to convert to; Vox Studio cannot guess this. If you don't feel like entering this every time you can preset this as a default using the Tools/Defaults menu. The extension to the output filenames can be the same as the input filename, it can be removed or another one can replace it.

- The sampling frequency can modify the sample rate of the file, i.e. the number of signal measurements performed per second. According to the Nyquist theorem the highest frequency component in any sound file cannot exceed 1/2 the frequency at which the file is sampled. So, when a file is down-sampled (resampled at a lower sampling rate) Vox Studio automatically filters the file to remove the components above 1/2 the sampling frequency before doing the re-sampling. Naturally, when a file is up-sampled (resampled at a higher sampling rate), there is no way Vox Studio can add frequency components that are not present in the file (or have been removed by previous down-sampling). In other words, down-sampling will necessarily result in a file with a lower bandwidth, but up-sampling will not increase the file's frequency content. Thus, when you do a down-sampling followed by a reverse up-sampling the resulting file will not be the original. If that is the intent, you should reload the original file from disk instead. Downsampling irremediably deteriorates a file and should always be the last thing you do to a file. You can select any of the "standard" telephony or multimedia sample rates from a list, but you can also input a custom "non-standard" sample rate if you need to.

While converting you can also adjust leading and trailing silence, improve intelligibility, normalize the sound amplitude, or filter the sound. These options can be selected using a tabbed section of the dialog box.

- **Trim leaders and trailers:** allows easy production of prompt files with uniform leading and trailing blanks. What used to be a time-consuming manual editing task has now become as easy as selecting a button with your mouse. The rest is done for you automatically.

Automatic silence adjustment is a threshold-activated process and thus requires spotless, clean recordings. If you have background noise in your recordings, Vox Studio may incorrectly detect the beginning and end of sound in your file.

The dialog box allows you to select the threshold level Vox Studio uses to detect the difference between silence and non-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 based on the threshold you selected. 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 threshold level can be set either as a percentage of the maximum on-screen amplitude or as a threshold value in dB, dBm or dBV.

The silence duration itself is defined in the Tools/Defaults/Convert Active menu. A reasonable typical value would be 300 milliseconds. Note that the Leader/Trailer option is actually capable of adding silence to your file if you request more leading or trailing silence than your file actually contains!

If the voice files you are trimming are not recorded in optimal conditions it may be very useful to perform a "Center" or a "Normalize" operation on your file before you apply the "Leader/Trailer" command. This has the effect of "flattening" the low frequency or DC signal background on the zero line and "pumping up the signal" which makes it so much easier to perform a good threshold detection on your file.

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a signal that just reaches the floor and the ceiling of the display. As a result, if you have recorded speech and you select a maximum average level of energy of say 70% in the Normalize command you can get sound spikes whose amplitude will very briefly be clipped at 100%. For voice recordings, this happens on very short sounds like "T", "P" and "K" which are explosive sounds anyway and where amplitude saturation does not matter very much. This technique allows recording at as high a level as possible, with best possible telephony quality. A similar technique is used when recording with a recording device that has a recording meter and adjusting the level to remain below 0 dB most of the time (green area lights) but allowing some very short spikes to exceed 0 dB (red area lights briefly).

The best practical approach is to calibrate your Vox Studio recording and playback setup once and for all by fine-tuning your recording settings. You can use the monitor function and the graphical display function to do that. Your signal should go from time to time in the red range on the Monitor VU-meter and should use at least 75% of the available amplitude range shown on the graphical waveform display screen. It does not matter too much if your signal peaks sometimes and briefly (we mean briefly) hits the ceiling or the floor. This usually will happen for utterances with T, P and K where a little clipping does not matter so much. You should select a preferred normalization factor that gives best volume and quality results when played back to the specific telephony card you use. Once that is set, your settings should usually never change unless you change the target telephony system. Remember that the Normalize option is there to correct minor variations between recordings; it is not there to correct grossly clipped or grossly under-recorded signals. You should regularly check if you use nearly the full dynamic range you have available by looking at your recorded messages in the graphical display window. Note: when Normalize is selected, a Center function is in fact performed automatically on the signal prior to amplitude normalization.

Finally, although this functionality is described elsewhere in the chapter describing the calibration tool, remember that Vox Studio's dBV or dBm display can be calibrated so that the dB levels you see on-screen actually match the dB levels that the same signal will generate on the target telephone line.

- **Center around zero:** automatically re-centers the sound file around the zero baseline to eliminate DC offsets caused by your sound card, and possibly by very-low-frequency interference. Note: when a Normalize command is issued, a Center command is, in fact, performed automatically on the file prior to amplitude normalization.

Depending on the quality of your sound card, and your recording setup, it may also be very advantageous to perform a centering operation before doing leader/trailer work (which requires rather precise threshold detection). Vox Studio does not automatically perform a centering operation before every leader/trailer command but you may select this option.

You can perform a quick visual check using the graphical waveform display in the main window. Record a file with nothing but silence using your onboard sound card. Save the file and then reload it into Vox Studio. You should see a clean horizontal line exactly superimposed on the zero baseline. If you see two lines your hardware does generate significant offset. If you see only one line the offset generated is small enough that you can disregard it. If you see green rubbish (grass) on the baseline your recording setup picks up unwanted noise (which you will certainly hear in the final recording).

If you manipulate files that were recorded on another workstation, it is a good idea to load the files first and visually check them for a possible DC shift.

DC offsets are very disturbing whenever you need to amplify a signal or do threshold detection. Also too much DC offset can generate audible clicks at the beginning and end of a file.

- **Intelligibility filter:** consists of a Clarity filter which corrects the muffled sound effect often obtained when down-sampling voice files and a Boost option which produces variable signal amplification in

order to increase the perceived voice energy content of the recorded file. The Intelligibility Filter can be selected only while doing a Convert Active, Convert Current or a Batch Conversion operation. Intelligibility Filter options are set in the defaults section for the conversion functions. Usually a "weak" clarity filter and "no boost" is the best choice. Use trial-and-error to select the options that suit your voice files best.

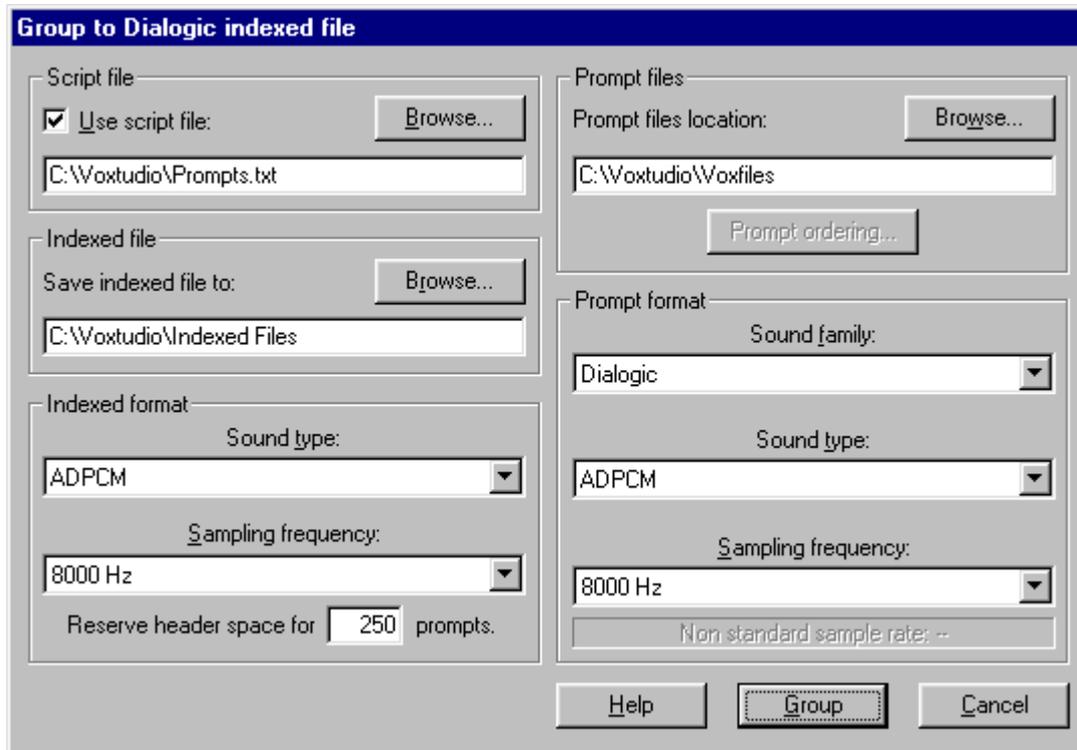
- **Low- and high-pass filter:** allows individual selection of either a high-pass (low-cut) or a low-pass (high-cut) filter. For the high-pass and low-pass filters the -3 dB cutoff frequencies can be chosen with a 1 Hz resolution. According to the Nyquist theorem the highest frequency component in any sampled file cannot exceed 1/2 the frequency at which the file is sampled. Audible "aliasing" errors occur whenever the Nyquist theorem is overlooked. It makes little sense to select a cutoff frequency above 1/2 the sampling frequency of the file.
- **DTMF filter:** removes signal components at and near the frequencies that correspond to one group of the DTMF frequencies. This harsh treatment effectively removes talk-off problems, but also degrades the sound quality somewhat. Use this only on files that really do cause talk-off problems and cannot be rerecorded. The DTMF filter has 3 strengths you can choose from: weak attenuates 10 dB, medium attenuates 20 dB, and strong attenuates 30 dB. Use the weakest filter that solves your talk-off problem.

The Batch Convert command can also be activated by clicking on  in the toolbar section.

5.6.4. Group to Dialogic indexed file

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

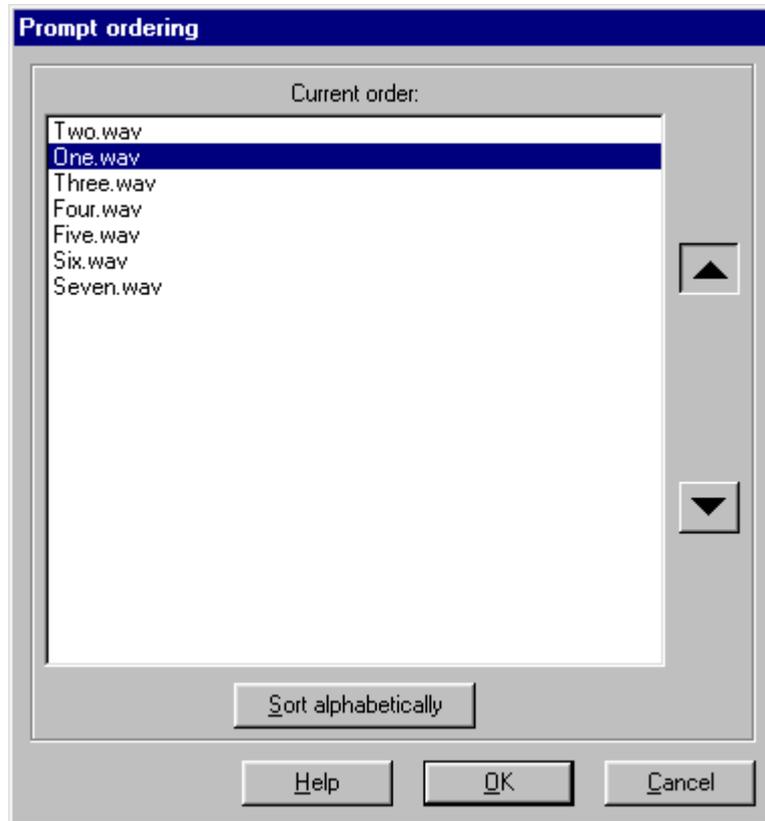
One of the very nice features in this version of Vox Studio is that you can group any type of single-prompt file into a grouped (indexed) file. If a format conversion is necessary, Vox Studio does the conversion transparently on-the-fly. It is thus possible, for instance, to start with single-prompt ".wav" files and group them into a ".vap" file as a single operation. There is no need to start with Dialogic files to obtain a Dialogic indexed (grouped) file. To be grouped input files have to be single-prompt files; they cannot themselves be multi-prompt grouped files.



Vox Studio ".vap" indexed files are compatible with the standard Dialogic ".vap" indexed files as produced by the manual voice editors sold by Dialogic and their resellers.

A script file is used to tell Vox Studio which stand-alone files to regroup into an indexed file. This script file is identical in content to the script file used for the prompter and tape loader commands. See elsewhere for a detailed definition of the [script file format](#).

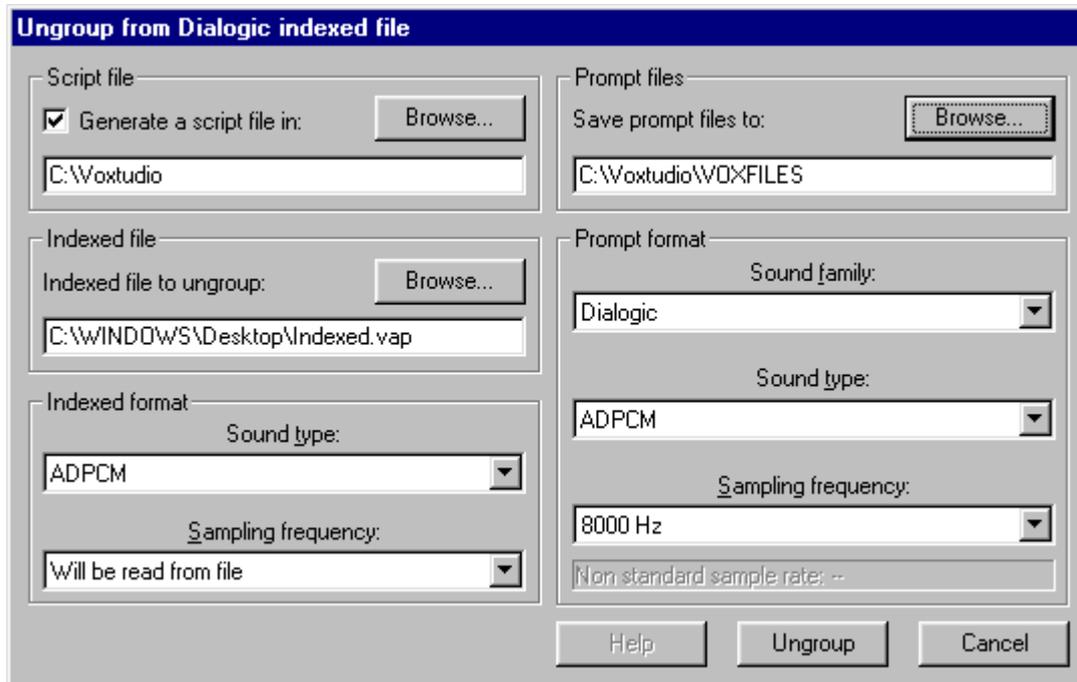
If the files to be grouped are chosen manually (i.e. you are not using a script file for the grouping operation) you can press the Prompt Ordering button to open the prompt ordering window and select the physical order under which you want the files to appear in the indexed file.



Pressing the up- and down-arrow buttons moves the selected file up or down in the list. You can keep those buttons pressed to move a file over a long distance in the list.

5.6.5. Ungroup from Dialogic indexed file

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

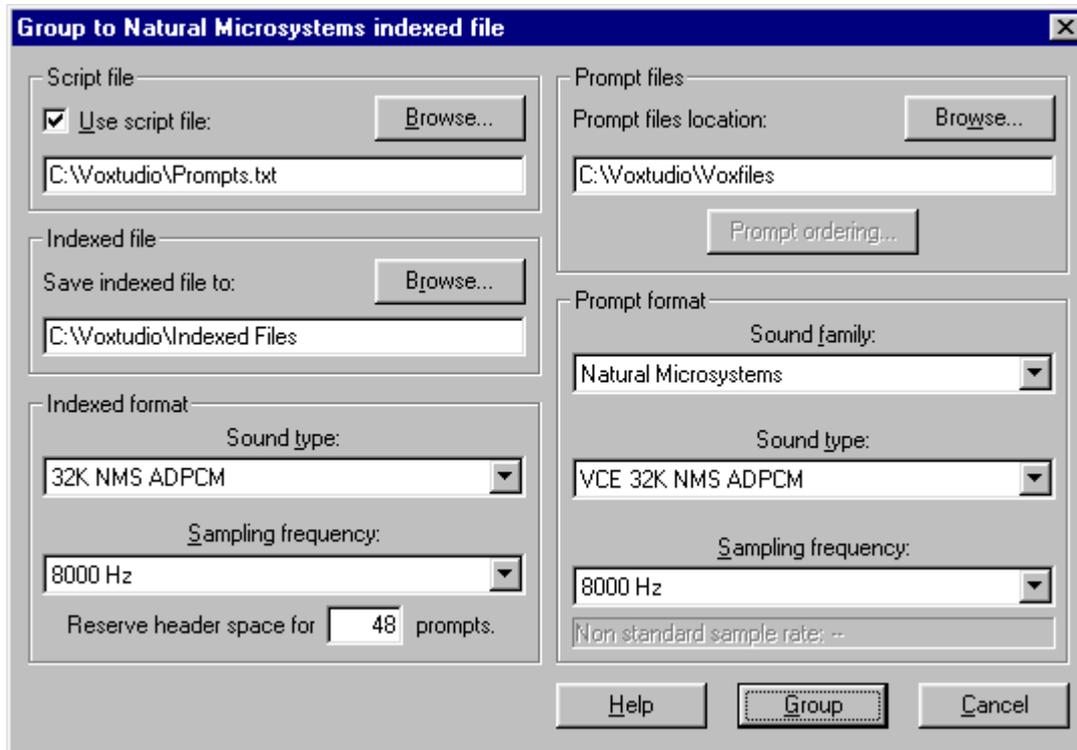


Optionally, a script file can be produced for seamless regrouping of the stand-alone components after manipulation.

5.6.6. Group to NMS indexed file

Group to NMS indexed file allows concatenation of several stand-alone telephony voice files (often with a ".vox" or ".vce" extension) into one single indexed NMS telephony voice file (often with a ".vox" extension).

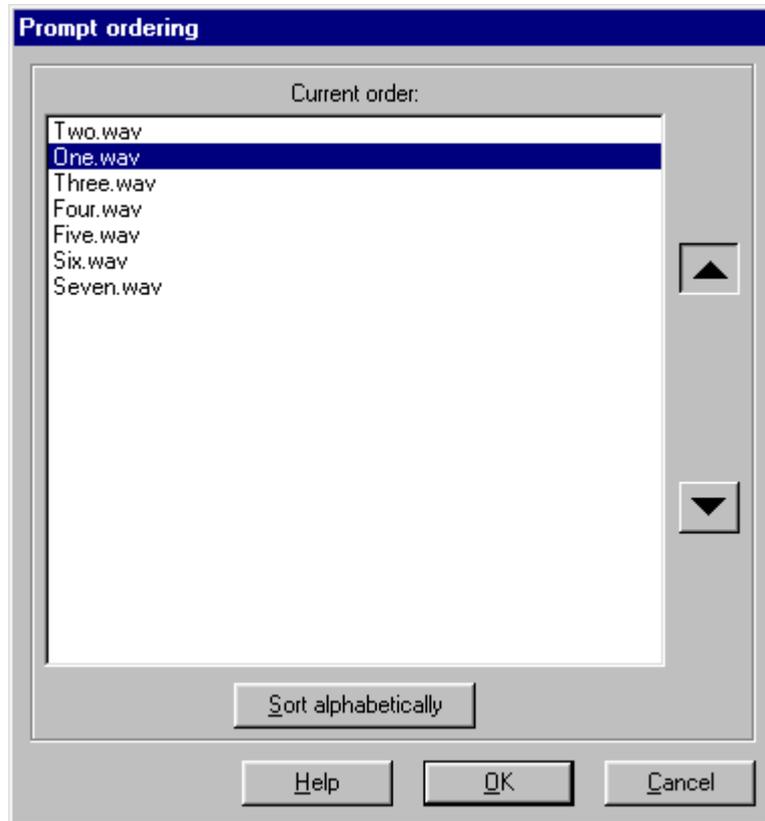
One of the very nice features in this version of Vox Studio is that you can group any type of single-prompt file into a grouped (indexed) file. If a format conversion is necessary, Vox Studio does the conversion transparently on-the-fly. It is thus possible, for instance, to start with single-prompt ".wav" files and group them into an indexed ".vox" file as a single operation. There is no need to start with NMS files to obtain an NMS indexed (grouped) file. To be grouped input files have to be single-prompt files; they cannot themselves be multi-prompt grouped files.



Vox Studio ".vox" indexed files are compatible with the standard NMS ".vox" indexed files as produced by the manual voice editors sold by Natural MicroSystems and their resellers.

A script file is used to tell Vox Studio which stand-alone files to regroup in an indexed file. This script file is identical in content to the script file used for the prompter and tape-loader commands. See elsewhere for a detailed definition of the [script file format](#).

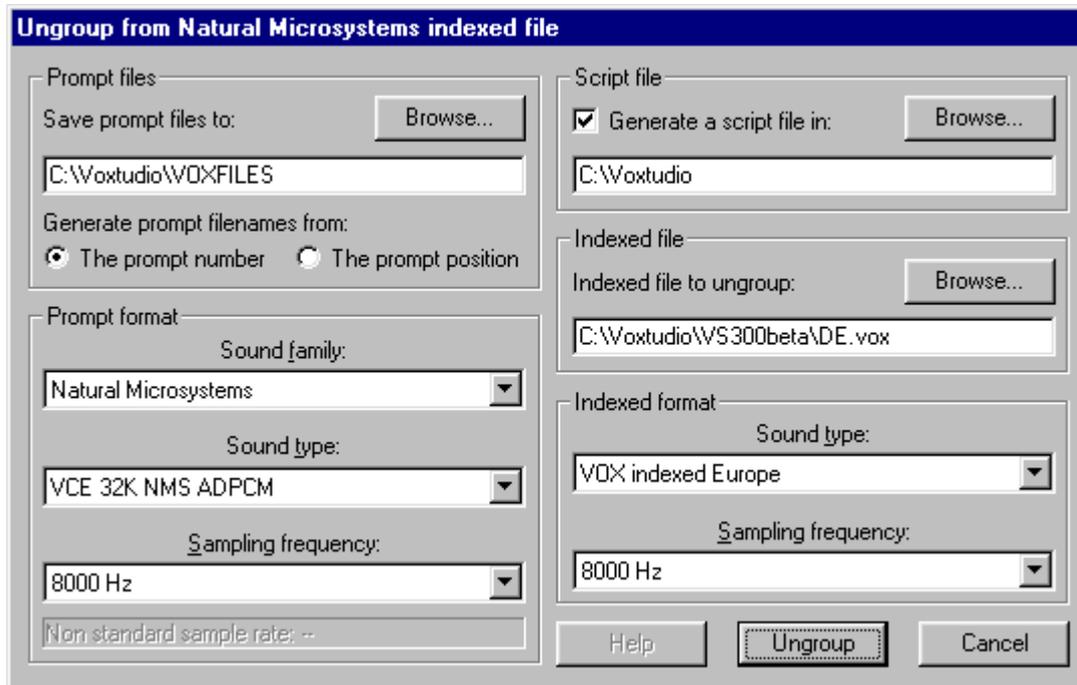
If the files to be grouped are chosen manually (i.e. you are not using a script file for the grouping operation) you can press the Prompt Ordering button to open the prompt ordering window and select the physical order under which you want the files to appear in the indexed file. The prompt index number inside the indexed file will be the same as the sequence number chosen here.



Pressing the up- and down-arrow buttons moves the selected file up or down in the list. You can keep those buttons pressed to move a file over a long distance in the list.

5.6.7. Ungroup from NMS indexed file

Allows expansion of a NMS indexed telephony file (usually ".vox") into its stand-alone voice file components (usually ".vox" or ".vce").



Optionally, a script file can be produced for seamless regrouping of the stand-alone components after manipulation.

5.7. Tools menu

This menu hosts the commands to record a single file, play a whole bunch of files, set the recording sensitivity, generate strings of DTMF digits, and access other, external programs.

- [Generate tones](#)
- [Detect tones](#)
- [Monitor Sound Input](#)
- [Play](#)
- [Record](#)
- [Play a list](#)
- [Program options](#)
- [Program defaults](#)
- [Sound devices](#)

5.7.1. Generate

Vox Studio currently generates DTMF tones only. Other telephony tones will be added in future versions.

- [Generate DTMF](#)

5.7.1.1. Generate DTMF

The DTMF tone generator has 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. Simply clicking on the corresponding keys on-screen generates a tone sequence. One can also add pause commands (represented in the display by a comma) in the DTMF sequence. The tones will be inserted at the current cursor position in the active file. If no file was active in Vox Studio when this command was selected, a new empty prompt will be created to contain the DTMF tones.

The default length of the individually produced DTMF tones and silences is programmable in the Tools/Defaults/DTMF menu.

Naturally, as one would expect, higher sample rates produce more accurate DTMF tones, and these are detected more accurately. Use the highest possible sample rate to generate DTMF tones. Do not generate DTMF tones in files with sampling rates as low as 6 KHz; they will usually not be accurate enough.

Do not forget to check the defaults settings for DTMF generation in the Tools/Defaults/DTMF menu. In particular, make sure you choose an output level that does not conflict with the regulatory requirements of your telephony provider. Usually an output level of around -9dB will be adequate.

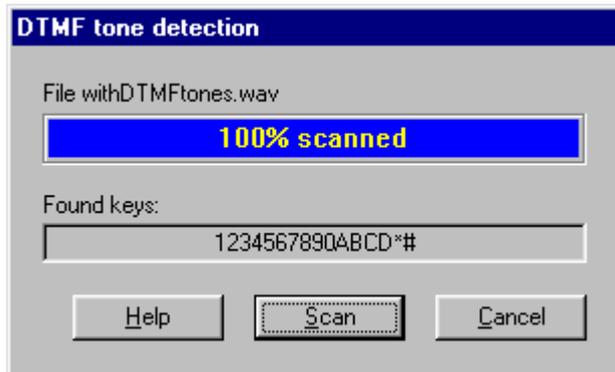
5.7.2. Detect

Vox Studio currently detects DTMF tones only. Other telephony tones will be added in future versions.

Detect DTMF

5.7.2.1. Detect DTMF

The DTMF tone detector scans your currently active file for DTMF tone pairs and shows if and which DTMF signals were detected in the file. Vox Studio detects all 16 DTMF signals; the detected digits include 1234567890, ABCD, * and #.



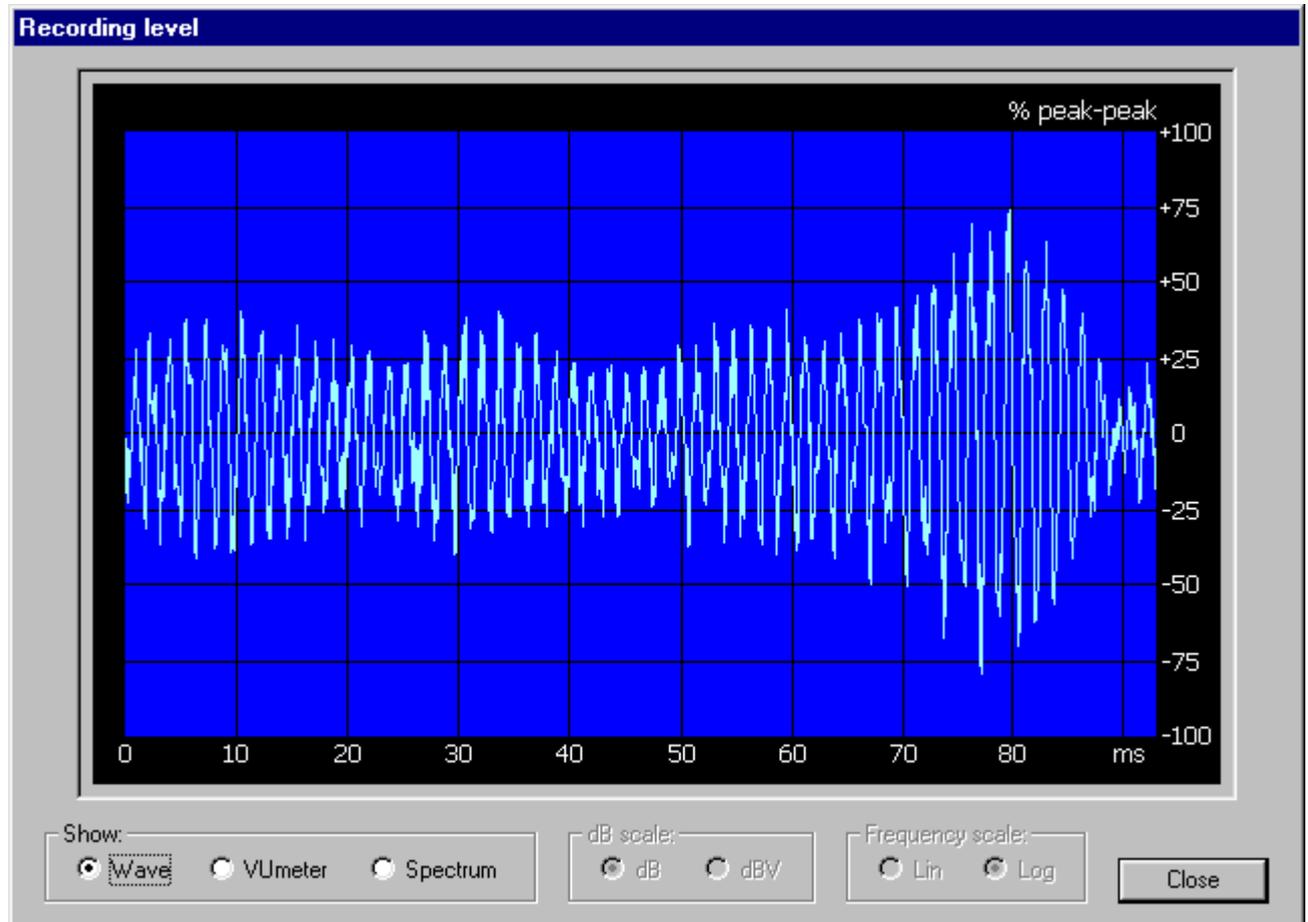
It is very important to set the detection threshold at a realistic value using the Tools/Defaults/DTMF menu. To give you some indication, a "must detect" level of -24dB and a "may not detect anymore" level of -32 dB are realistic values in telephony. Typically one is not allowed to place DTMF tones with very high amplitudes on a telephone line (above -9 dB for instance) so it is useless to try and detect DTMF tones above that value or below the minimum threshold value.

More productivity features based on this DTMF detection capability will be added in future versions of Vox Studio.

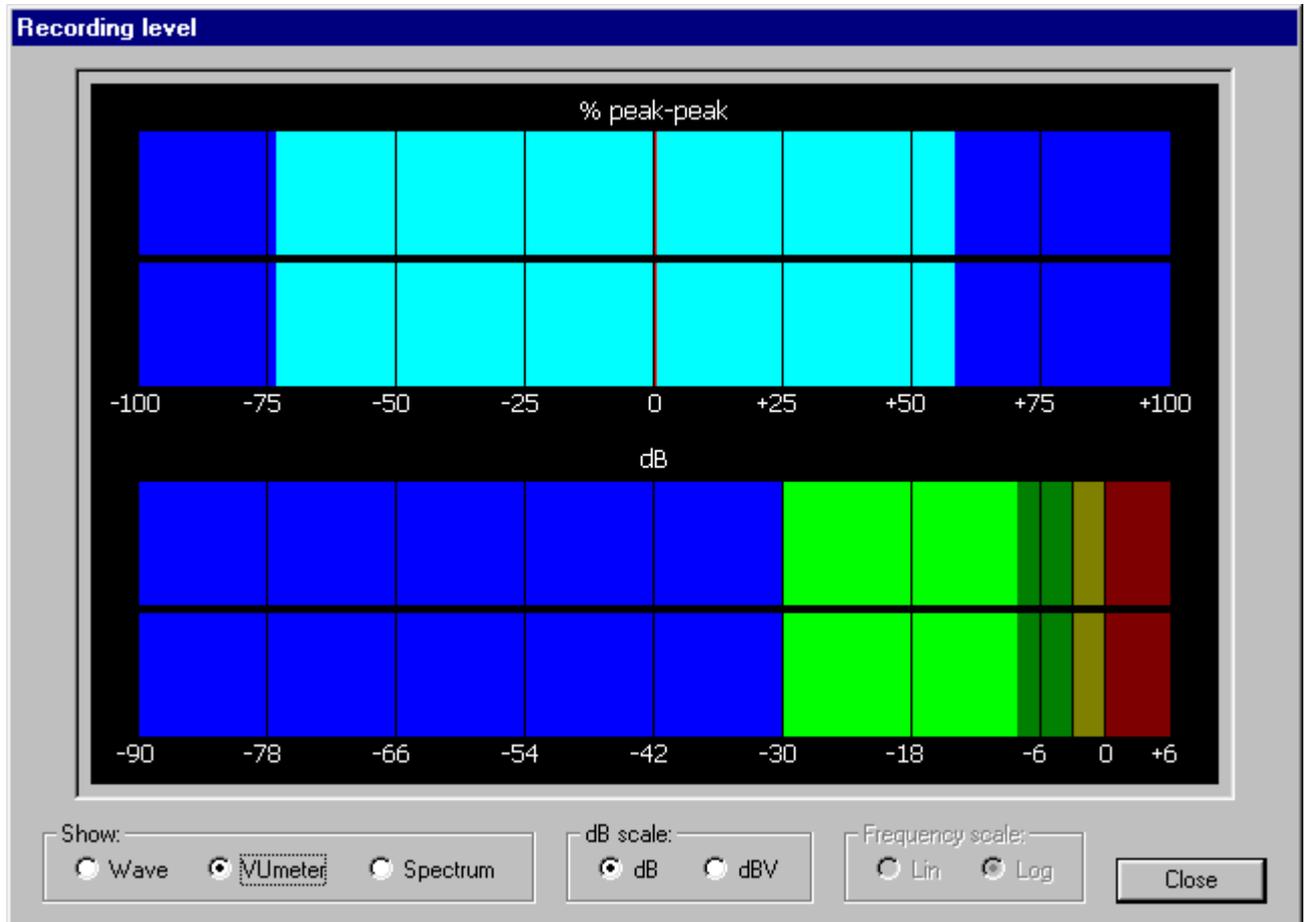
5.7.3. Monitor Sound Input

The recording level input monitor is useful during the early stages of the setup and tuning of your recording installation. There are three views: the waveform display, the VU-meter and a spectral display of the input signal. What you see is not what you have recorded but rather the direct signal input of your sound card.

The Waveform view shows a linear representation of the input signal and looks like this:



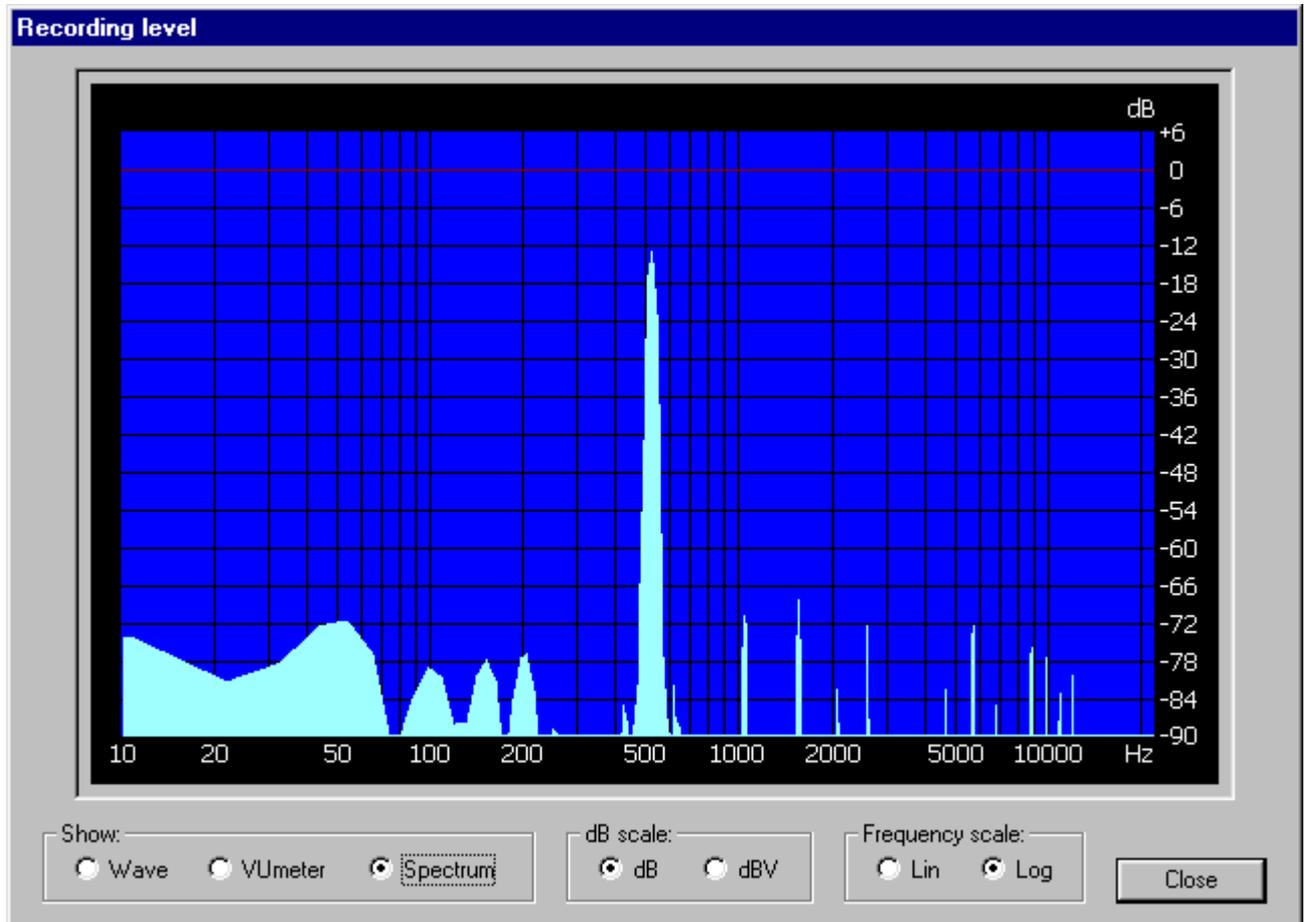
The VU-meter view shows a logarithmic (decibel) representation of the input signal and looks like this:



The VU-meter looks like the one used on professional tape recorders. It will allow you to calibrate the recording sensitivity of your multimedia sound card. While speaking normally into your microphone, tune the sensitivity with the volume adjustment utility provided by your card manufacturer. Make sure the visual indication goes only occasionally into the red area (risk of signal clipping). Also make sure you are not always recording at too low a level (blue area). Low-level recordings require extra amplification later in the process and because this process amplifies both the desired signal and the background noise, this could cause unnecessary background noise in your recordings. A good compromise recording level is when the visual indication spends a lot of time in the green and yellow areas and sometimes goes very briefly into the red one.

Use the volume control applet that came with your sound card (this is card dependent) to adjust the input sensitivity such that the monitor color bar usually stays in the green or yellow areas when you speak into the microphone. Short peak bursts in the red area are of no importance. Avoid staying in the blue area for too long; you would be recording at a too low level.

The spectrum view shows the frequency content of the input signal. It is useful to detect background noise or other anomalies. It looks like this:



The spectral content can be viewed in linear amplitude or dB energy mode, and the frequency axis can be linear or logarithmic.

The Monitor Sound Input command can also be activated by clicking on player button  under the graph pane.

5.7.4. Play Selection

This command is only enabled if you have indeed selected part or the entire active signal in the graph window. When activated this command plays the selected portion of the signal.



This command is also available from the  play button on the toolbar at the bottom of the graph pane: if a selection is active, that selection will play, if no selection exists the whole file will play when pressing the play button.

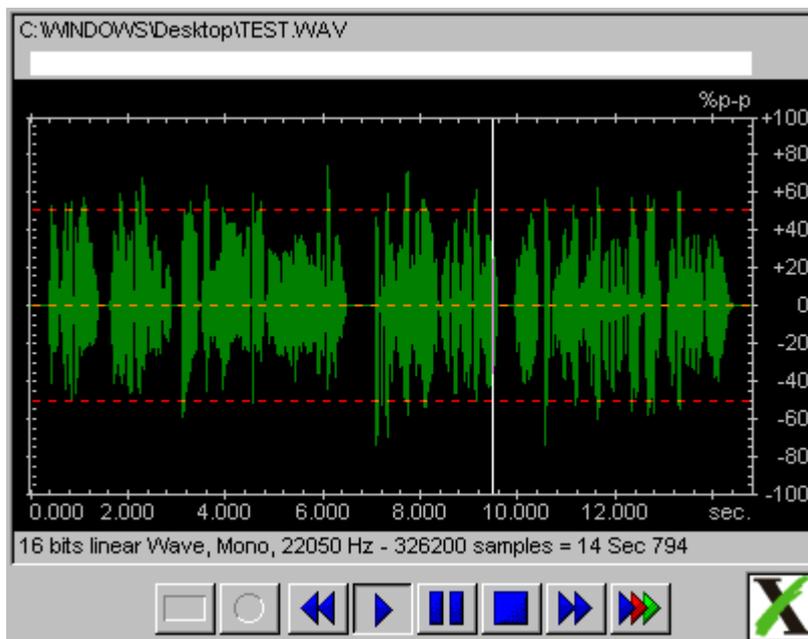
5.7.5. Play Window

This command is only enabled if you have already zoomed-in on part or the entire active signal in the graph window. When activated this command plays the visible portion of the signal.



This command is also available from the  play button on the toolbar at the bottom of the graph pane: if a zoom is active, that button will play the visible window if no selection exists, otherwise the selection will play when pressing the play button.

5.7.6. Play All



This command plays the whole active file. During play a moving cursor shows progress through the file.

This command is also available from the  play button on the toolbar at the bottom of the graph pane: if no selection is active and no zoom-in has been made, that button will play the whole sound file. Playback starts at the beginning of the file if no cursor has been positioned in the file, otherwise playback starts at the cursor position.

5.7.7. Play All But Selection

This is the opposite of the play selection command. It plays everything as one single sound but skips the selected part of the file if any selection exists. This allows you to listen to what a file would sound

like if you were to delete the selected portion.

5.7.8. Pause

This command pauses playback. Selecting the command again restarts playback where it had paused.



This command is also available from the  pause button on the toolbar at the bottom of the graph pane. Pressing the pause button pauses playback. Pressing it again restarts playback where it had paused.

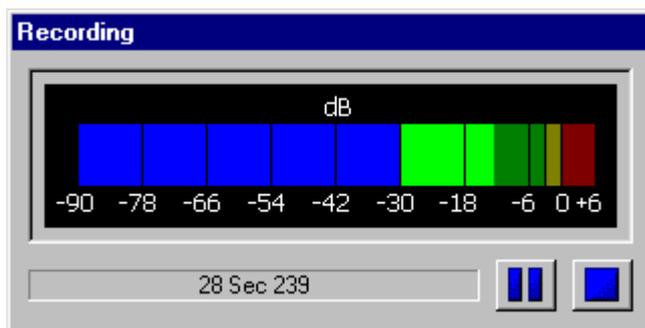
5.7.9. Stop

This command terminates playback.



This command is also available from the  stop button on the toolbar at the bottom of the graph pane. Pressing the stop button ends playback.

5.7.10. Record

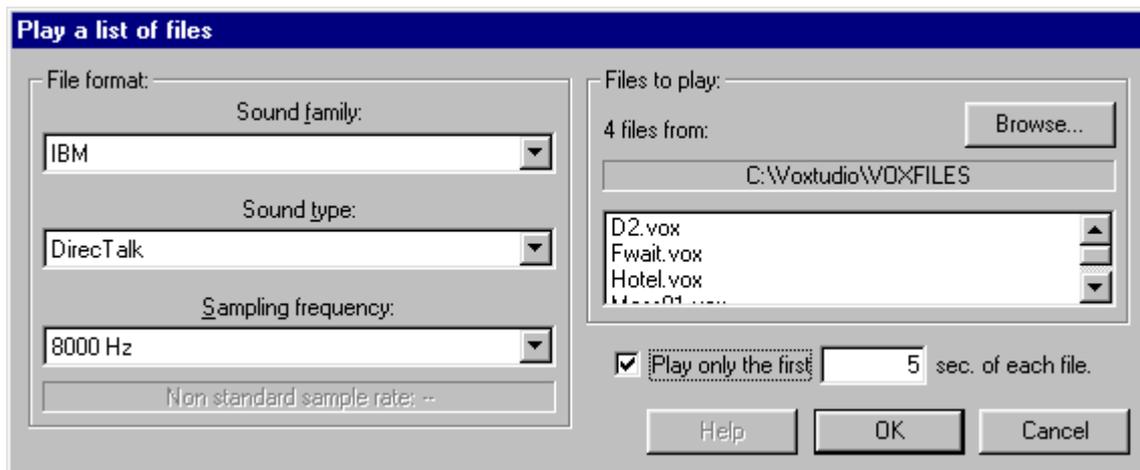


Pressing the red  record button opens a simple graphical window. From there you can pause or stop the current recording. Before starting a recording you should define the sampling rate (6 to 48 KHz) of the sound you wish to record. If an active window exists the recording will be inserted in the active file. If no active window exists, Vox Studio will ask you what sample rate you want to use and will create a new empty window for you. A sample rate of 11 or 22 KHz is usually a good compromise between quality and disk space.

After recording a prompt you can immediately play it back, edit it, or save it in the format of your choice. As the save operation allows you to define file coding, sample rate and resolution you can thus convert your recording into any telephony format known to Vox Studio. Once a prompt is saved, Vox Studio is ready for a new recording.

5.7.11. Play a list of files

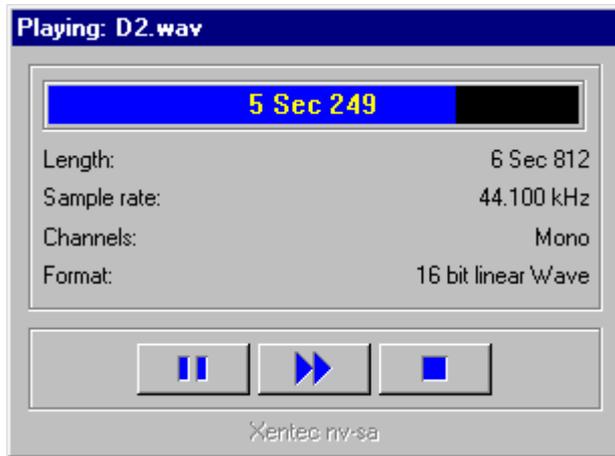
This command opens a selection Dialog box. Choose the format of the files to be played (they should all be in the same format). Use the browse button to select, in the sequence in which you want them played, all the files that need to be played. The usual Windows shortcuts of Click, Shift-Click, Ctrl-Click can be used to select a range of files or separate individual files. You can also type *.* wild characters in the browsing window to select all files in a folder. The maximum number of files that can be selected and played with a single "Play a list of files" command depends on the length of the file names: the total length of all file names added together should not exceed 65,000 characters, so that if you use "normal" filenames you can play a lot of files indeed.



Tell Vox Studio what type of file you want to play. If you select a headerless format you will have to give additional coding and sample rate information. If not, Vox Studio will read this information automatically from the file headers.

You can click a selection button so that the function only plays the first so many seconds of each file. This length is adjustable.

The following window appears when you press OK:



All files then automatically play one after the other. You can check which file is currently playing by looking at the title bar. Pressing the "Skip" button skips to the next file in the list. You can also pause or end playback.

It is thus possible to scan through hundreds of files very rapidly, irrespective of file format, as long as all the files are of the same format. There is even an option to only play the first "n" seconds (let us say 3 seconds) of every selected file for quick identification or checking purposes.

The display indicates the file length in seconds and milliseconds, the sampling rate in KHz, the recording format used and the stereo/mono status (for ".wav" files).

Some file formats, like ".wav" for instance, have information headers at the beginning of the files. For those files it is usually not necessary to indicate to Vox Studio what the precise recording parameters are for each file, as Vox Studio will read this from the file itself. For other file formats Vox Studio will prompt you for the file-family (Dialogic or CCITT for instance), coding type (e.g. A-law or ADPCM) and sampling frequency (e.g. 8 KHz or 11 KHz). If your files are of a single file family which has (type information) headers then you can mix files with different sampling rates or resolutions in a single play list command. If not, all files will need to have the same characteristics as there is no way Vox Studio can guess this information when going from one file to another.

The same functionality is also available to an external file manager program (Windows Explorer for example). You can select and drag all the filenames you want to play and drop them onto the Vox Studio play list icon at the bottom of the graph window (the three colored arrows). A slightly different window now appears:



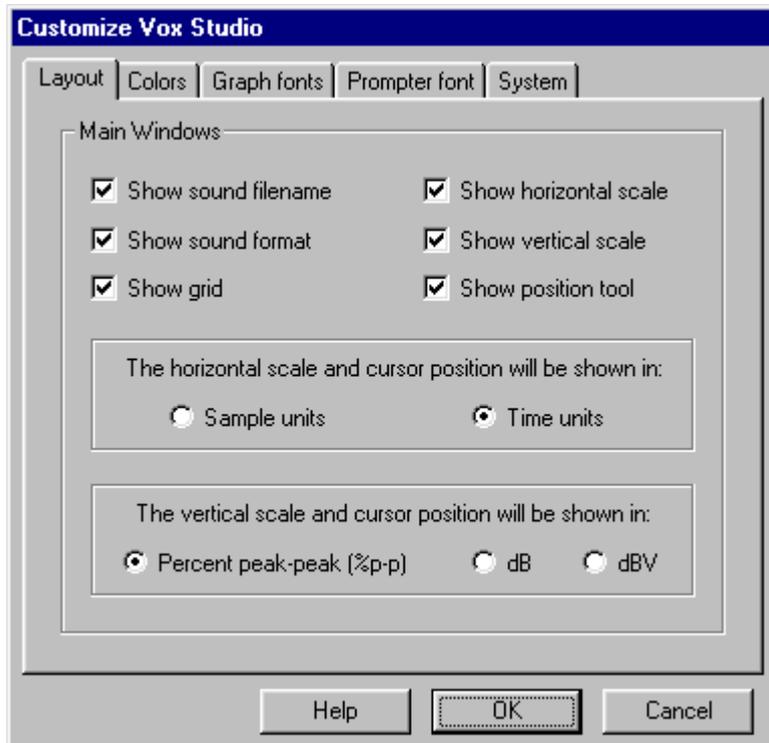
The functionality is similar to the standard method above, except that you will have chosen all the files to play in an external program instead of in Vox Studio.



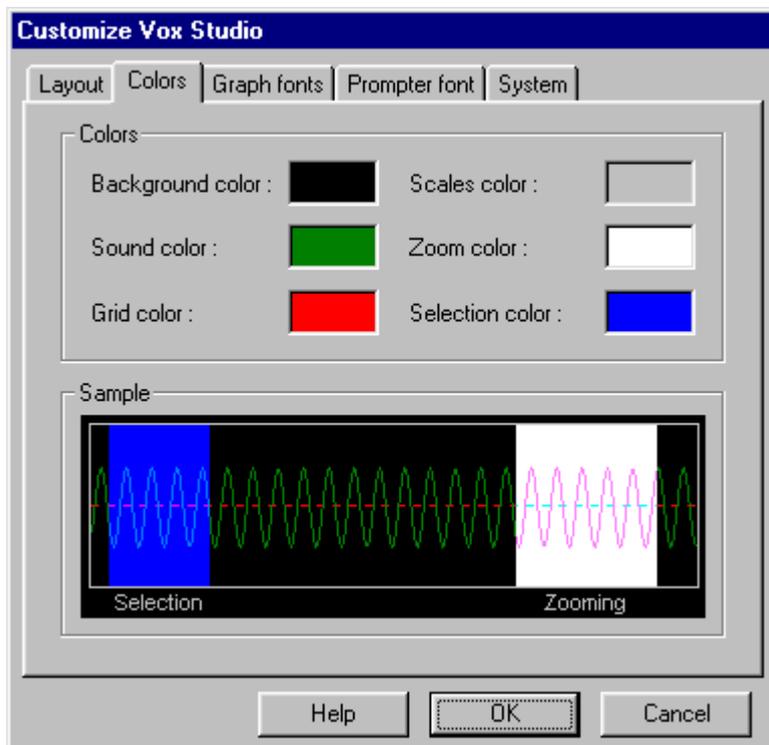
The Play List command is also available from the  pause button on the toolbar at the bottom of the graph pane. This same button can also be used as drop target for drag-and-drop playback.

5.7.12. Options

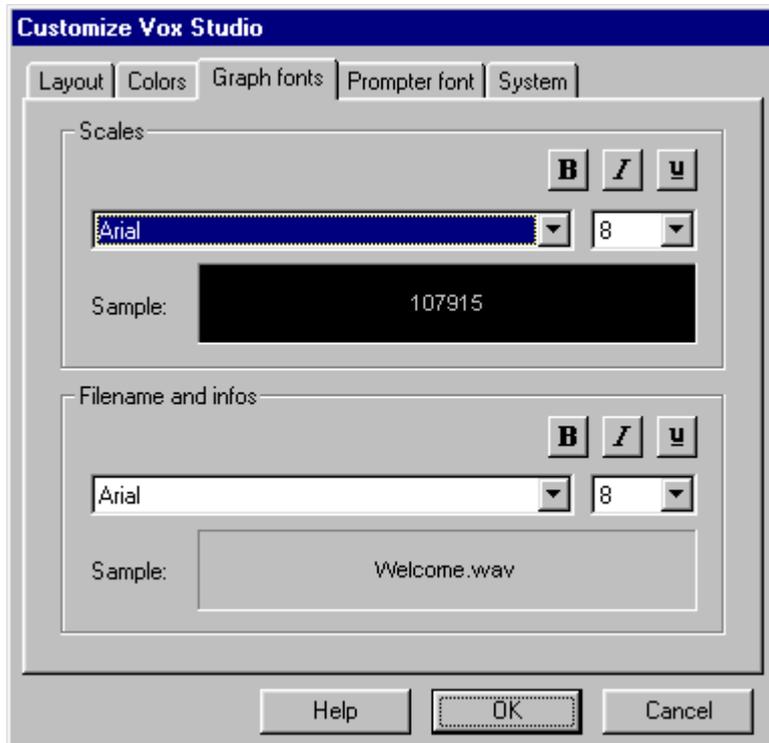
The Options command in the Tools menu customizes the way Vox Studio looks and acts.



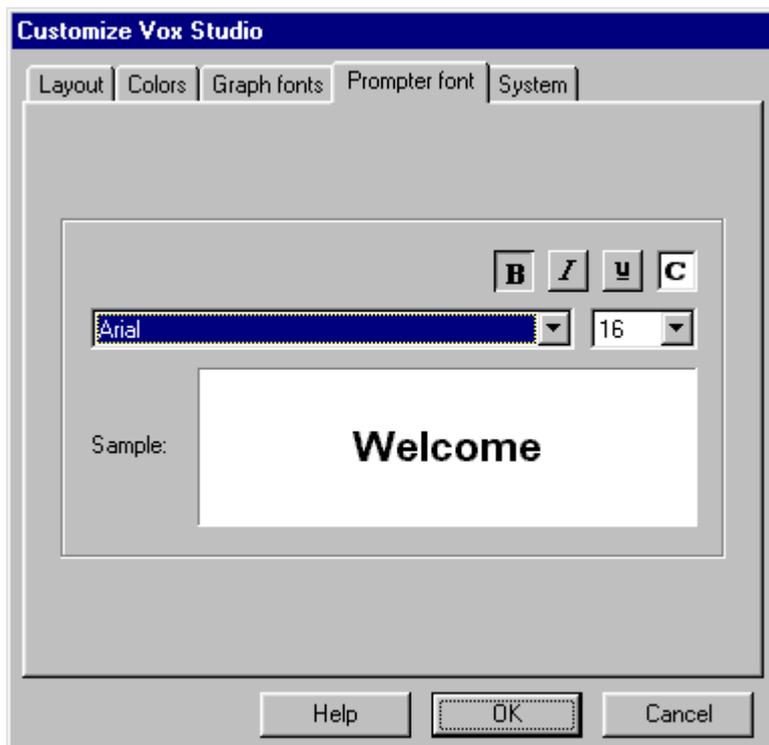
The layout tab allows customization of what information is displayed in the main windows and how the graphs are labeled and shown. The vertical scale setting (percent or decibels) is also available in the program contextual menu shown when right clicking on a graph window.



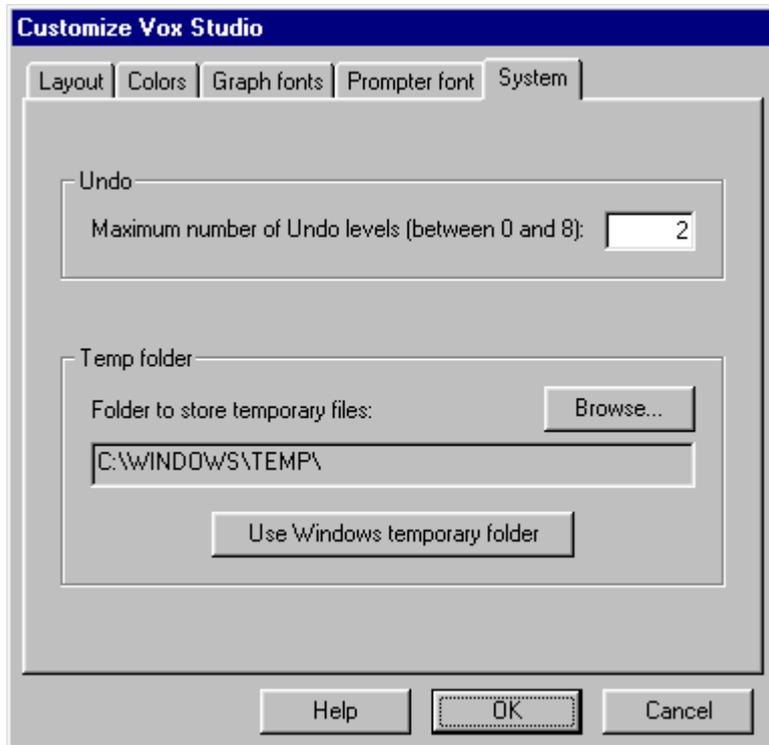
The colors tab allows customization of the colors used in the graph windows.



The Graph fonts tab allows selection of the character fonts used in the graph windows.



The Prompter font tab selects the character font used to display the message prompts on-screen when using the Prompter, Tape Loader and File Slicer.



The system tab allows selection of the maximum number of undo levels available in Vox Studio (the absolute maximum allowed is 8). This limit needs to be set very carefully because Vox Studio needs to keep a backup copy of altered sounds in order to undo the alterations. If you work with very large recordings, this can rapidly eat up hundreds of megabytes or even gigabytes.

The temporary folder is where Vox Studio stores its intermediate work files. This too can consume a lot of disk space if your sound files are large. You should select a temporary folder located on the disk partition that has the largest amount of free disk space. This may or may not be the disk where Windows itself keeps its temp folder.

5.7.13. Defaults menu

We very strongly suggest you browse through the various default options in the Tools/Defaults menu and set the defaults to your liking before you start using Vox Studio! Most of the support calls we receive are due to inappropriate settings in the Tools/Defaults menu.

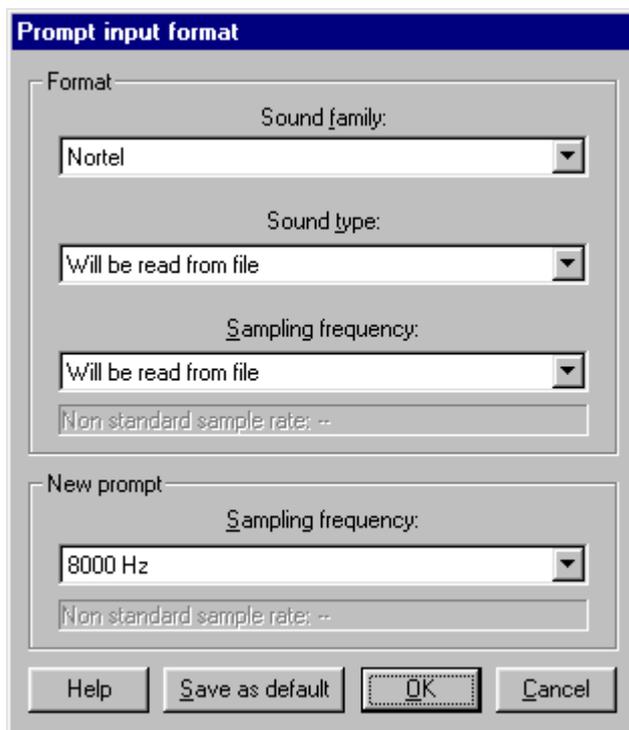
At the very least you should set the default working directories, file formats and extensions used for multimedia and telephony files, and you should also select sound card device drivers in the Tools/Sound Devices menu or Vox Studio will not work satisfactorily.

Most default-setting windows have both an "OK" and a "Save as default" button. Pressing "OK" uses the selected default for this session of Vox Studio. Pressing "Save as default" also stores the default value for use in later sessions of Vox Studio.

[Input Format defaults](#)
[Output Format defaults](#)
[Folder defaults](#)

[Play a List of Files defaults](#)
[Convert/Transform Active defaults](#)
[Batch Conversion defaults](#)
[Prompter defaults](#)
[Tape Loader defaults](#)
[File Slicer defaults](#)
[Group and Ungroup Dialogic defaults](#)
[Group and Ungroup NMS defaults](#)
[dBV and dBm Calibration](#)
[DTMF defaults](#)

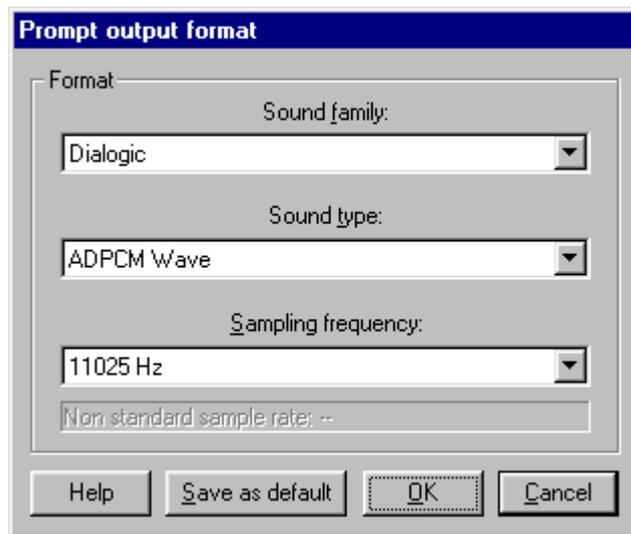
5.7.13.1. Input Format defaults



This command allows selection of the default source conversion format and file open format. This is the source format that will originally be selected in the dialog boxes associated with file conversion or file open operations. Vox Studio does detect the input format if the necessary information is in the file header; otherwise it presents the format entered here as the default in Dialog boxes.

If you plan to manipulate many files, all in the same format, it makes a lot of sense to select this format here as the default. If you do not, you will need more manual intervention when you use Vox Studio. Selecting the right default here will make working with Vox Studio much more productive.

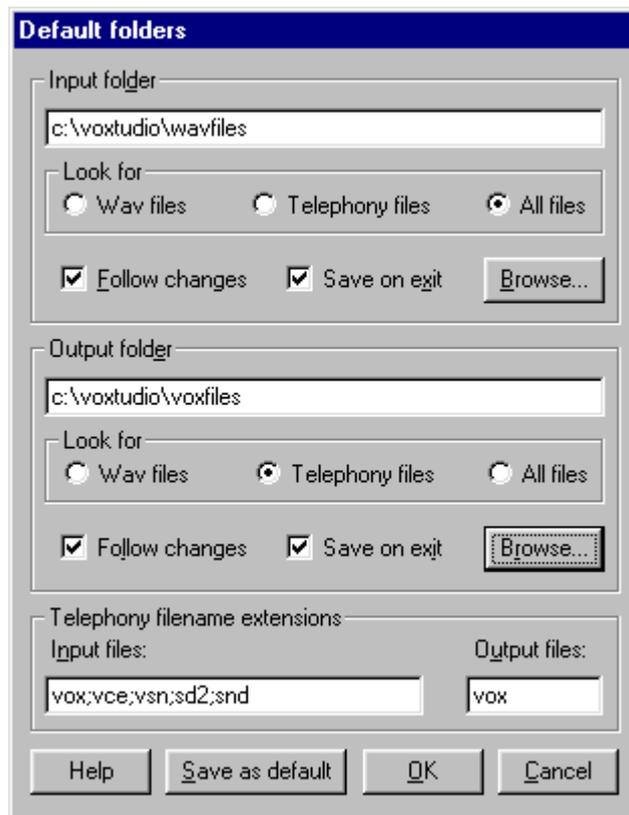
5.7.13.2. Output Format defaults



This command allows selection of the default format presented when saving sound files. This is the target format that will originally be selected by default when you open the dialog boxes associated with file saving operations.

If you plan to convert many files to the same format, it makes sense to select this format as the default. If you do not, you will need more manual intervention when you use Vox Studio. Selecting the right default here will make working with Vox Studio much smoother.

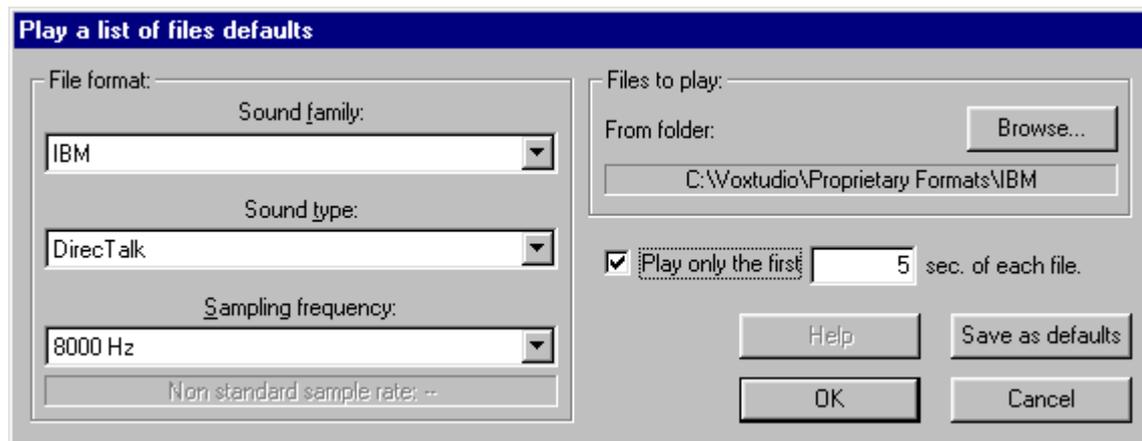
5.7.13.3. Folder defaults



This command allows selection of the default directories that will hold the sound files and telephony voice files you are going to use in Vox Studio. If you activate "follow changes", then, whenever you navigate to another directory in 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.

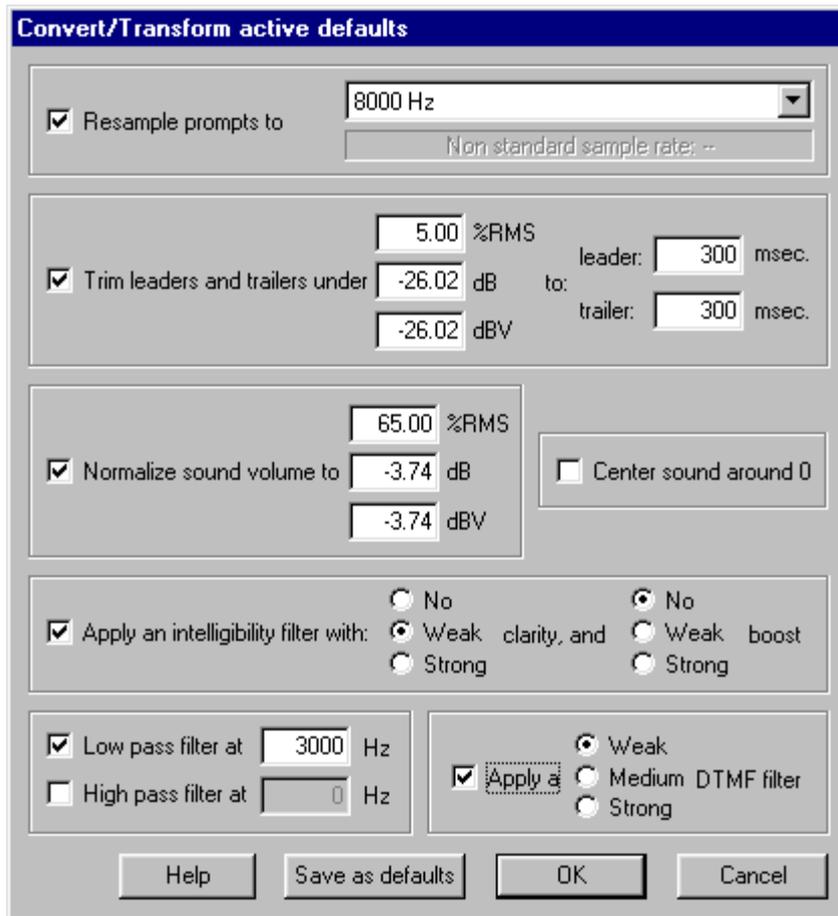
This command also allows selection of the usual filename extension(s) for opening or for saving the telephony voice files used by your specific voice applications. There is a separate input box for the default file name extension when saving telephony files. You can only enter one default extension for saving telephony files, but you can have several extensions for opening files. Enter the 3-letter extension(s) separated by semicolons and no blanks (for instance: vox;vce;vsu;sd2;snd). These extensions will be used when listing telephony files in selection boxes. If you want files without any extension, just enter a semicolon (;).

5.7.13.4. Play a List of Files defaults



Use this to select the format presented by default when playing a list of files in the "Tools/Play a list of files" command and in what folder to look for them by default. If you select your most-frequently-used format and location here, you will have less navigation work to do when playing a whole bunch of files later. You can also select here whether the default will include playing only the beginning of files, and how many seconds should be played.

5.7.13.5. Convert/Transform Active defaults



This is where you set the default values to be used in the Transform Active and Convert Active commands. The available default settings are:

- **Resample prompt:** set the default to modify the sample rate of the file, i.e. the number of signal measurements performed per second. The Nyquist theorem states that the highest frequency component in any sound file cannot exceed 1/2 the frequency at which the file is sampled; remember that when you set your defaults. You can select any of the "standard" telephony or multimedia sample rates from a list, but you can also input a custom "nonstandard" sample rate if you need to.
- **Trim leaders and trailers:** sets the defaults for trimming leading and trailing blanks.

The dialog box allows you to select the threshold level Vox Studio uses to detect the difference between silence and non-silence. The threshold level can be set as either a percentage of the maximum on-screen amplitude or as a threshold value in dB, dBm or dBV. Don't set the threshold lower than your background noise level, or nothing will work.

The default silence duration itself is also defined here. A reasonable typical value would be 300 milliseconds.

- **Normalize sound volume:** sets the default maximum sound energy for normalization. The selection can be done in % of the maximum on-screen amplitude, in dB or in dBm/dBV. Vox Studio does not measure peak AMPLITUDE, it measures peak ENERGY over a duration of several milliseconds

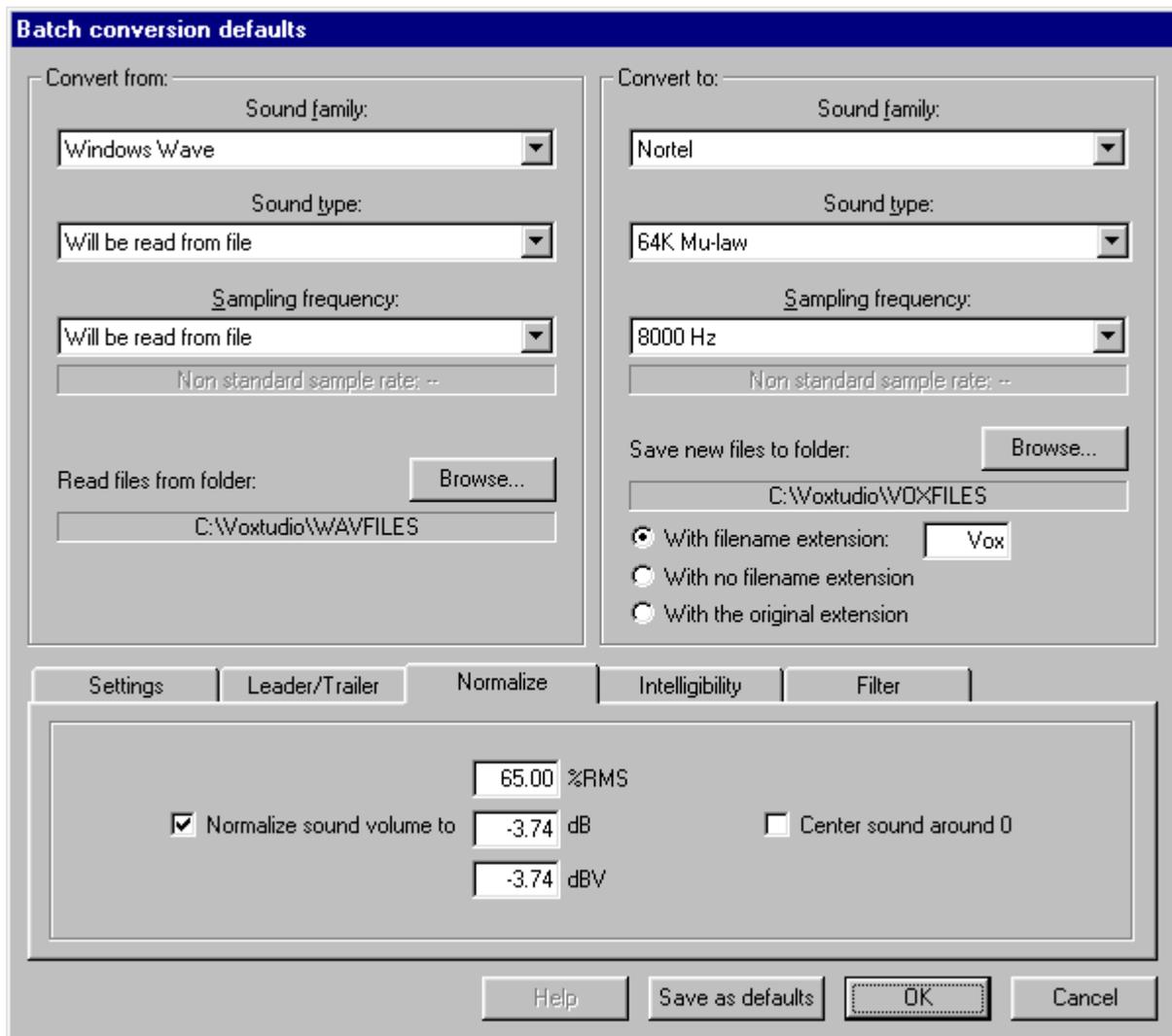
(about the duration of a spoken syllable). Usually a choice of about 65% of maximum energy will be fine.

Remember that you can also enter values in decibels and that Vox Studio's dBV or dBm display can be calibrated.

- **Center around zero:** if this is set here, then centering will be activated by default in the conversion dialog boxes. Disable it here if you usually will not use this feature.
- **Intelligibility filter:** if this is selected here, then the Intelligibility Filter option will be activated by default when initiating the Transform Active and Convert Active commands. You can also select the default Clarity Filter and Boost options here. Usually a "weak" clarity filter and "no boost" is the best choice.
- **Low- and high-pass filter:** if this is selected here, then the low- or high-pass filter options will be activated by default when initiating the Transform Active and Convert Active commands. For the high-pass and low-pass filters the -3 dB cut-off frequencies can be chosen with a 1 Hz resolution. Remember that according to the Nyquist theorem the highest frequency component in any sampled file cannot exceed 1/2 the frequency at which the file is sampled. You do not need to activate this option for sample rate conversions because Vox Studio does all the necessary associated filtering automatically.

DTMF filter: if this is selected here, then the DTMF filter option will be activated by default when initiating the Transform Active and Convert Active commands. The DTMF filter has 3 strengths you can choose from: "weak" attenuates 10 dB, "medium" attenuates 20 dB, and "strong" attenuates 30 dB. Use the weakest filter that solves your talk-off problems.

5.7.13.6. Batch Conversion defaults



- **Convert from/to:** You can select the default source directory for files to be converted and the default target directory for converted files. You can also select the default file types and the extension for the converted output files. These are the default values that will be presented when you open a batch conversion window.

You can set a default target sample rate for the converted file, i.e. the number of signal samples per second. The Nyquist theorem states that the highest frequency component in any sound file cannot exceed 1/2 the frequency at which the file is sampled; remember that when you set your defaults. You can select any of the "standard" telephony or multimedia sample rates from a list, but you can also input a custom "non-standard" sample rate if you need to.

A batch conversion log file "voxbatch.log" is saved in the same directory as the Vox Studio program itself during conversion. Rather than stop a batch command because an error occurred somewhere, Vox studio writes an error notice in this log file and continues to convert from there on. Always check the log file after a batch conversion.

- **Trim leaders and trailers:** sets the defaults for trimming leading and trailing blanks.

The dialog box allows you to select the threshold level Vox Studio uses to detect the difference between silence and non-silence. The threshold level can be set either as a percentage of the maximum on-screen amplitude or as a threshold value in dB, dBm or dBV. Don't set the threshold lower than your background noise level, or nothing will work.

The default silence duration itself is also defined here. A reasonable typical value would be 300 milliseconds.

- **Normalize sound volume:** sets the default maximum sound energy for normalization. The selection can be done in % of the maximum on-screen amplitude, in dB or in dBm/dBV. Vox Studio does not measure peak AMPLITUDE, it measures peak ENERGY over a duration of several milliseconds (about the duration of a spoken syllable). Usually a choice of about 65% of maximum energy will be fine.

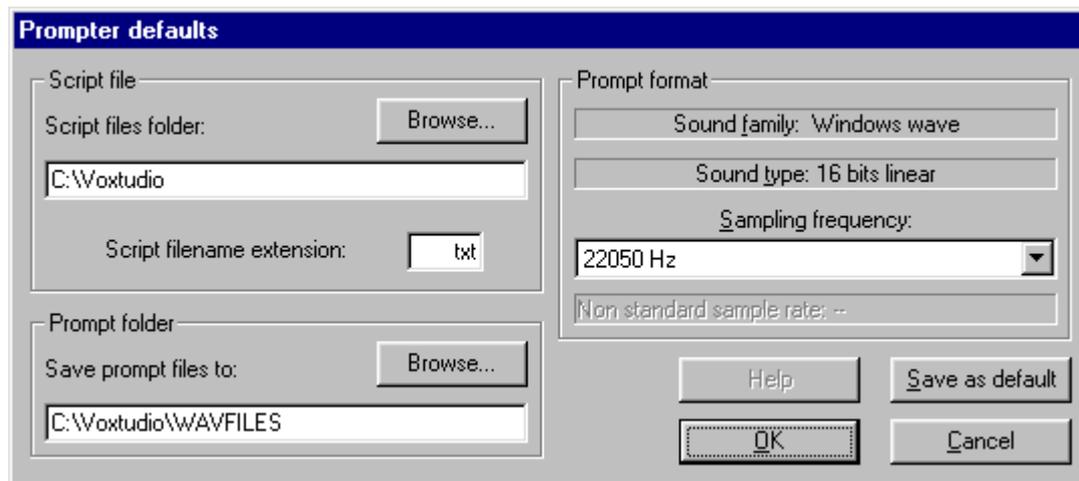
Remember that you can also enter values in decibels and that Vox Studio's dBV or dBm display capability can be calibrated.

If "center around zero" is selected here, then centering will be activated by default in the conversion dialog boxes. Disable it here if you usually will not use this feature.

- **Intelligibility filter:** if this is selected here, then the Intelligibility Filter option will be activated by default when initiating the Transform Active and Convert Active commands. You can also select the default Clarity Filter and Boost options here. Usually a "weak" clarity filter and "no boost" is the best choice.
- **Low- and high-pass filter:** if this is selected here, then the low- or high-pass filter options will be activated by default when initiating the Transform Active and Convert Active commands. For the high-pass and low-pass filters the -3 dB cutoff frequencies can be chosen with a 1 Hz resolution. Remember that according to the Nyquist theorem the highest frequency component in any sampled file cannot exceed 1/2 the frequency at which the file is sampled. You do not need to activate this option for sample rate conversions because Vox Studio does all the necessary associated filtering automatically.

DTMF filter: if this is selected here, then the DTMF filter option will be activated by default when initiating the Transform Active and Convert Active commands. The DTMF filter has 3 strengths you can choose from: "weak" attenuates 10 dB, "medium" attenuates 20 dB, and "strong" attenuates 30 dB. Use the weakest filter that solves your talk-off problems.

5.7.13.7. Prompter defaults



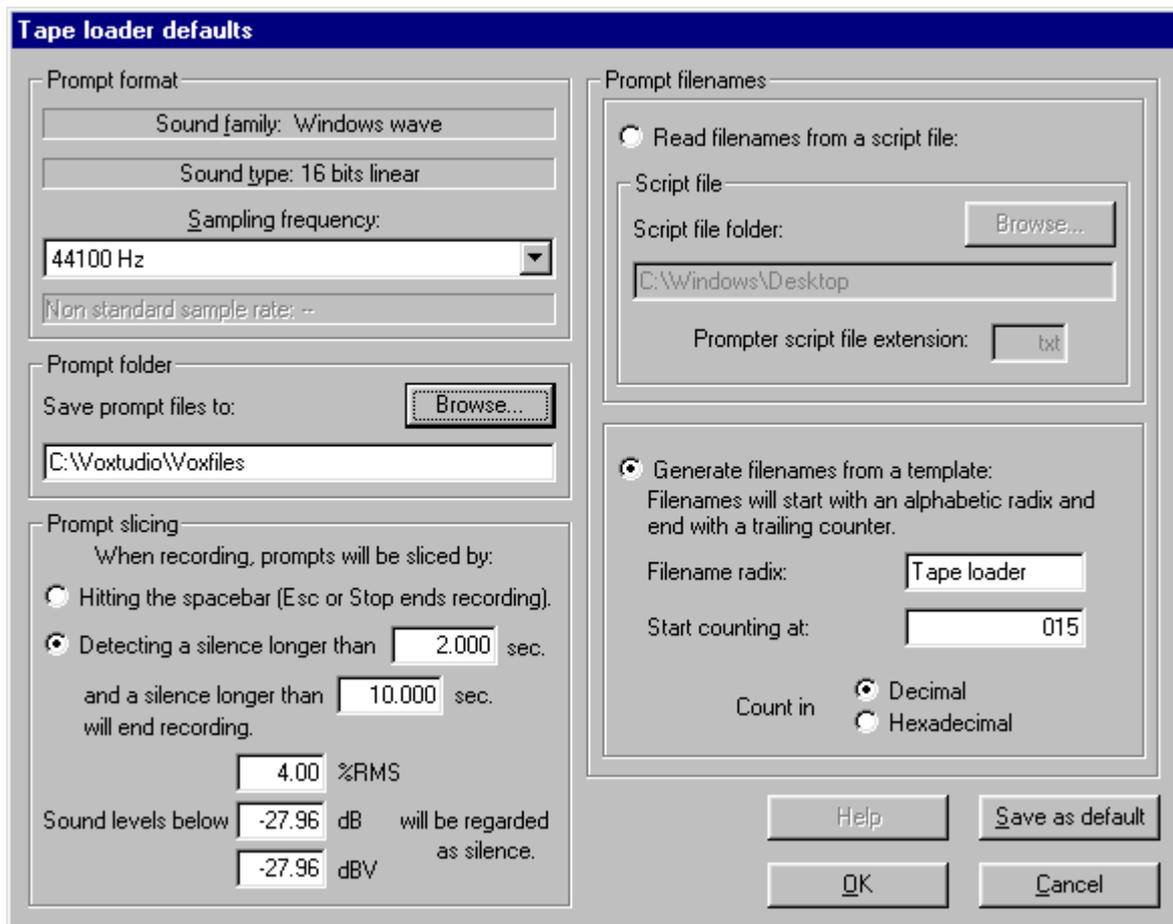
Here you define the default sample rate to be used for the Prompter recording sessions.

You can also define the directory where the recorded files will be saved by default during a Prompter session. If you don't enter a "save-to" path in the prompter defaults then the complete path with filename (without extension though; it is always ".wav") has to be entered in the script file itself. If a save-to directory is entered in the prompter defaults, only a filename (without extension) must be entered in the script file.

You can select the filename and location of the default script file used for Prompter sessions.

All these default options can be changed while in the Prompter itself, but having these set correctly here will facilitate your work.

5.7.13.8. Tape Loader defaults



This command allows you to select all the default operating parameters of the tape loader. These settings can be changed from the tape loader itself with the Parameters button.

Select how the filenames for the recorded messages will be created. You can select a script file, in which case the filenames will be taken from the script file itself. Caution: If you don't enter a "save prompt files to" folder in the tape loader defaults then the complete path (without extension though, it is always ".wav") has to be entered in the script file itself. If a save-prompt-files-to folder is entered in the tape loader defaults only a filename (without any extension) should be entered in the script file.

Instead of using a script file, you can also have Vox Studio generate filenames automatically. These will consist of a fixed alphabetic radix and a variable (incrementing) numerical suffix. The numeric part can be decimal or hexadecimal. For example if the starting radix is ivr and the trailing number is 1087, then filenames from ivr1087.wav to ivr9999.wav can be generated.

You can also define the default technique for the tape loader to detect the separation between recorded tape messages. You can tap the space bar yourself whenever you recognize the audible end of a prompt or you can have Vox Studio do this for you by detecting a settable number of seconds of silence. It would be a good idea to have 3 to 5 seconds of silence as a delimiter on your tape recordings. You can also define what length of silence represents the end of all tape recordings. This should be much longer than the longest separating silence on the tape. You can also set the threshold level Vox Studio will use to make the difference between silence and non-silence. The threshold can be set in % or in decibels.

Finally, select the sample rate of the recorded ".wav" files (from 6 to 64 KHz) and the disk folder where these will be stored.

5.7.13.9. File Slicer defaults

This command allows you to select all the default operating parameters of the File Slicer. These settings can be changed while working with the File Slicer itself by pressing the Parameters button.

Choose how the filenames for the sliced messages will be created. You can select a script file, in which case the filenames will be taken from the script file itself. Caution: If you don't enter a "save prompt files to" folder in the tape loader defaults then the complete path (without extension though, it is always ".wav") has to be entered in the script file itself. If a save-prompt-files-to folder is entered in the file slicer defaults only a filename (without any extension) must be entered in the script file.

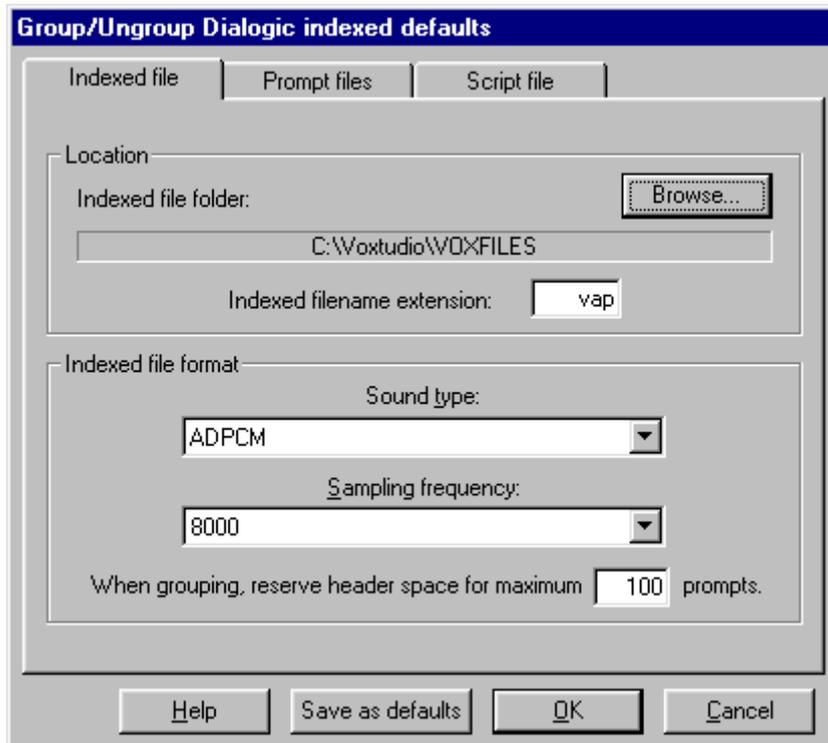
Instead of using a script file, you can also have Vox Studio generate filenames automatically. These will consist of a fixed alphabetic radix and a variable (incrementing) numerical suffix. The numeric part can be decimal or hexadecimal. For example, if the starting radix is FileToCut and the trailing number is 0050, then filenames from FileToCut0050.wav to FileToCut9999.wav can be generated.

You can also define the default number of seconds of silence Vox Studio uses to separate the prompts concatenated within the large ".wav" file. It would be a good idea to have 3 to 5 seconds of silence as a prompt delimiter in your large concatenated ".wav" recordings. You can also define what

length of silence represents the end of all tape recordings. This should be much longer than the longest separating silence on the tape. The threshold for silence detection can be set in % or in decibels.

Finally, select the disk folder where the new files will be stored. The file format for the new files will be the same as the file format of the input file.

5.7.13.10. Group and Ungroup Dialogic defaults



Select the default directory where the indexed files typically reside.

Define the default extension to be used for indexed filenames. The ".vap" extension is usually used for Dialogic indexed files, while the stand-alone voice files often have the ".vox" extension.

Specify the default coding and sample rate to be used for indexed files.

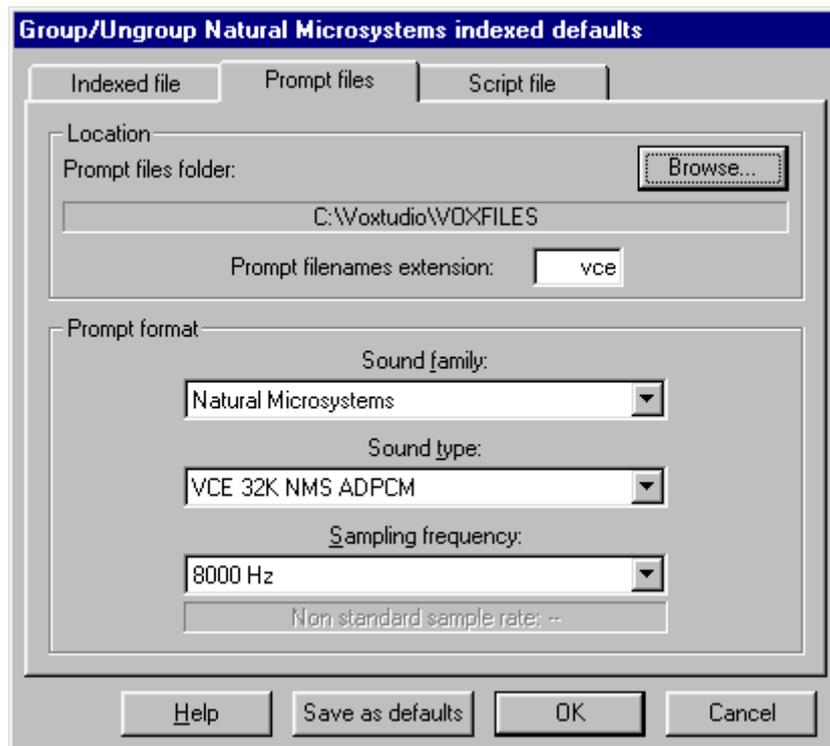
Specify the default maximum number of prompts per indexed file: header space for that many files will be reserved in the file. Choose this number somewhat larger than the largest number of prompts you typically have in an indexed file.

Select the default directory where the separate prompt files typically reside.

Specify the default format (family, coding and sample rate) to be used for the separate prompt files.

Select the default directory where the script files typically reside and the script file extension (usually ".txt")

5.7.13.11. Group and Ungroup NMS defaults



Select the default directory where the indexed files typically reside.

Define the default extension to be used for indexed filenames. The ".vox" extension is usually used for NMS indexed files, while the stand-alone voice files often have the ".vce" extension.

Specify the default coding and sample rate to be used for indexed files.

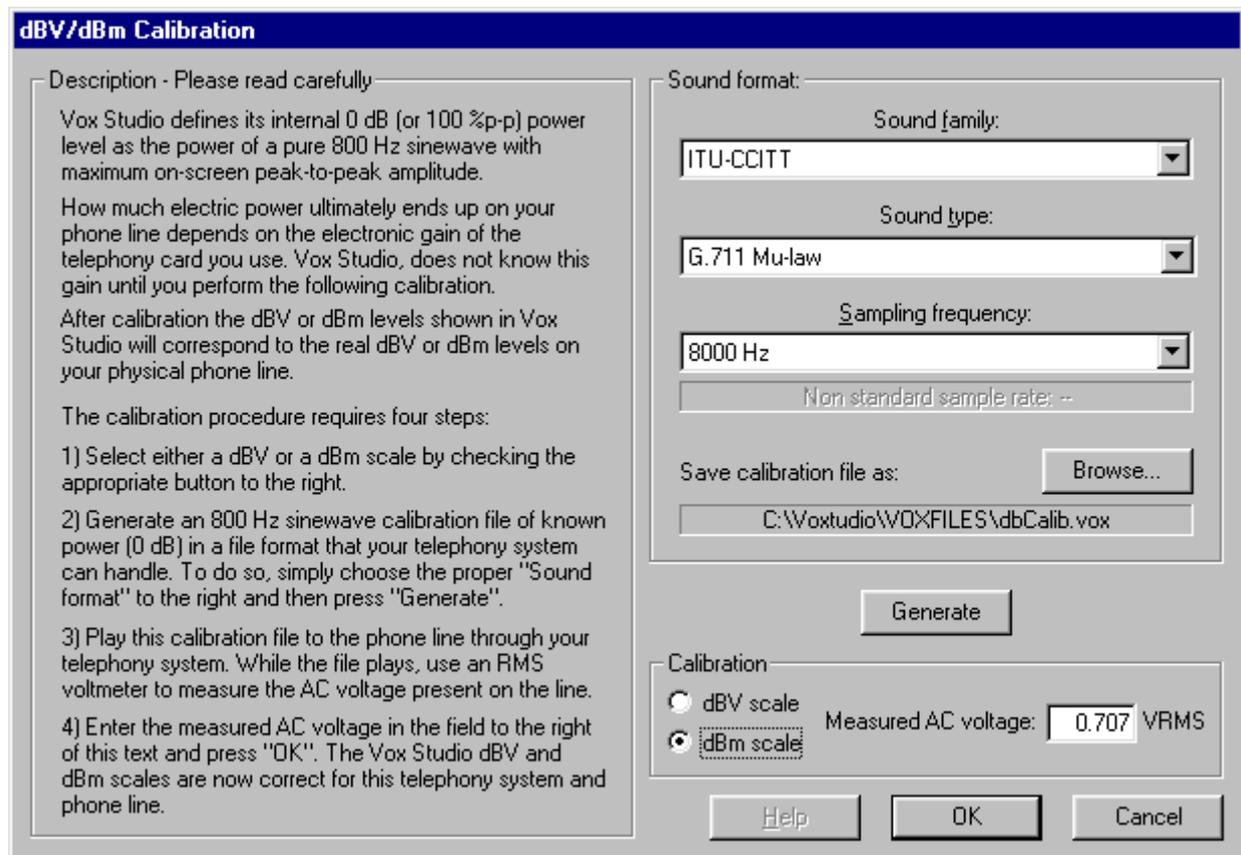
Specify the default maximum number of prompts per indexed file: header space for that many files will be reserved in the file. Choose this number somewhat larger than the largest number of prompts you typically have in an indexed file.

Select the default directory where the separate prompt files typically reside.

Specify the default format (family, coding and sample rate) to be used for the separate prompt files.

Select the default directory where the script files typically reside and the script file extension (usually ".txt")

5.7.13.12. dBV and dBm Calibration



This command allows precise calibration of your dBm and dBV displays and settings. The procedure outlined above (on the left of the dialog box) should be followed each time voice prompts have to be prepared for a different hardware platform (with different electrical levels on the phone line from the same digital file). In other words, this procedure calibrates Vox Studio to the electrical characteristics of the hardware on which the voice files will be used. Once this calibration is done the dBV and dBm levels used in Vox Studio will correspond to the real-world dBV and dBm levels on your target system.

5.7.13.13. DTMF defaults

	Low freq. group	High freq. group	Both groups	
Amplitudes:	25.12	25.12	50.24	%p-p
	-12.00	-12.00	-8.99	dB
	-12.00	-12.00	-8.99	dBV

Low/High freq. groups relative amplitudes:
 Independent Equal High: + 2dB

Tone duration: 150 msec.
 Inter-tone duration: 50 msec.
 Pause duration: 200 msec.
 Leading silence duration: 50 msec.
 Trailing silence duration: 50 msec.

Detection parameters:

Minimum tone amplitude: 6.31 %p-p
 -24.00 dB
 -24.00 dBV

Maximum twist level: 200.00 %p-p
 6.02 dB
 6.02 dBV

Minimum tone duration: 40 msec.
 Minimum pause duration: 30 msec.

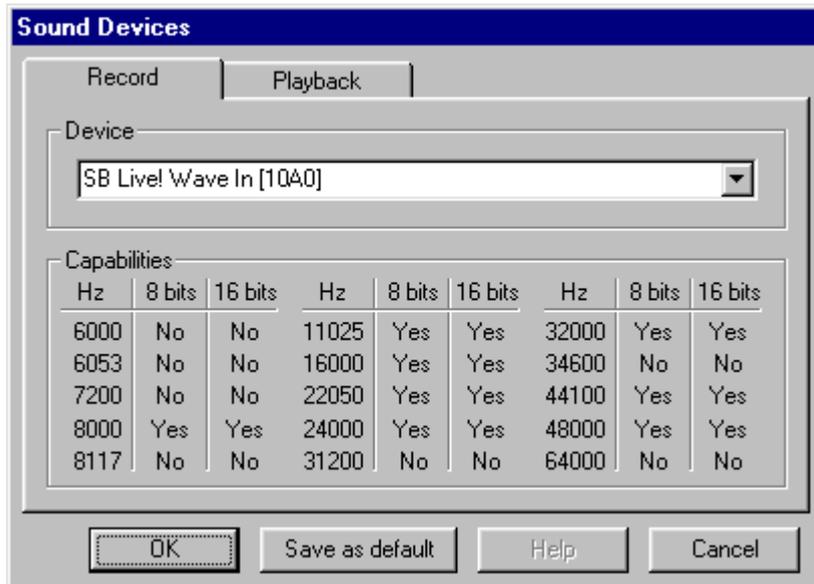
Buttons: Help, OK, Cancel

DTMF tones consist of a mixture of tones from two frequency groups. For DTMF generation these settings define the relative amplitude of both frequency groups. The amplitudes can be set to be equal for both groups, to be twisted by 2dB (the high frequencies are more attenuated on some phone lines, so they are sometimes set 2dB higher than the low frequency group), or to be set independently for both frequency groups.

The duration and separation of tone bursts can be set, as well as the duration of pauses (silences), and of file leading and trailing silences (silent padding before and after the tone sequence).

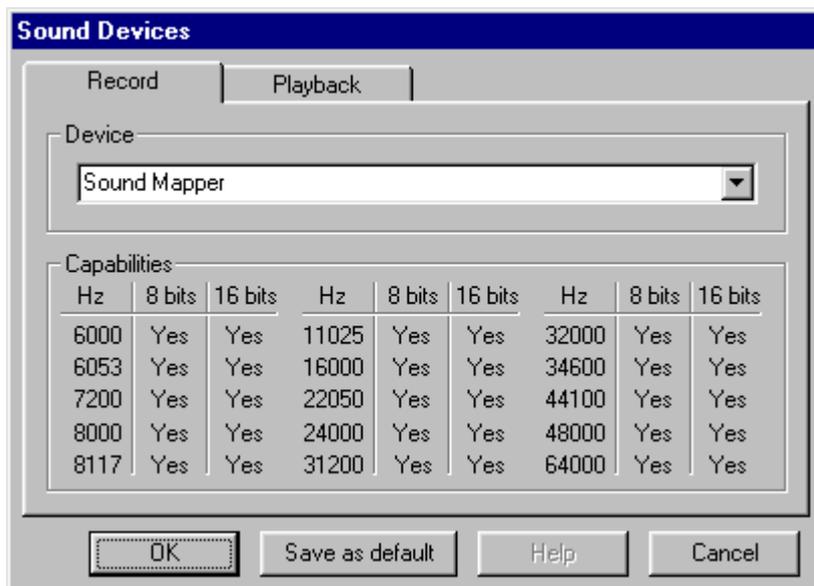
For DTMF detection, the minimum tone amplitude setting defines the threshold level above which DTMF tone pairs are considered to be valid DTMF digits. You can also set the maximum difference in amplitude (twist level) between the high frequency tone and the low frequency tone for DTMF signals to be considered as valid. Finally, you can set the minimum duration for a DTMF signal to be detected as valid and the maximum allowable gap (loss of signal) in such a DTMF signal. If there is a gap in the tone below this value then only one tone is seen. If the gap is above this value, then two DTMF tones are seen. It is important to set all these values at realistic levels or DTMF detection will be totally meaningless.

5.7.14. Sound Devices



This command allows selection of one sound input device and one sound output device from the sound devices currently installed under Windows. It is possible to have several sound cards and drivers installed in a system; this is where you select which ones Vox Studio should use. Vox Studio also shows you the capabilities of your sound card and driver here; this is useful for diagnostics. You can individually select which device to use for input and which for output.

You will not be able to record or play any file with Vox Studio until you have selected an installed sound input and output device. You can, however, use Vox Studio to do conversions only, without an input or output sound device.



If you have the Microsoft Sound Mapper installed (it is just software usually installed by default with Windows sound support) it may be a good idea to select it because Sound Mapper allows you to play 16-bit files even if your sound card is only an 8-bit card. Naturally, you will obtain better results with a

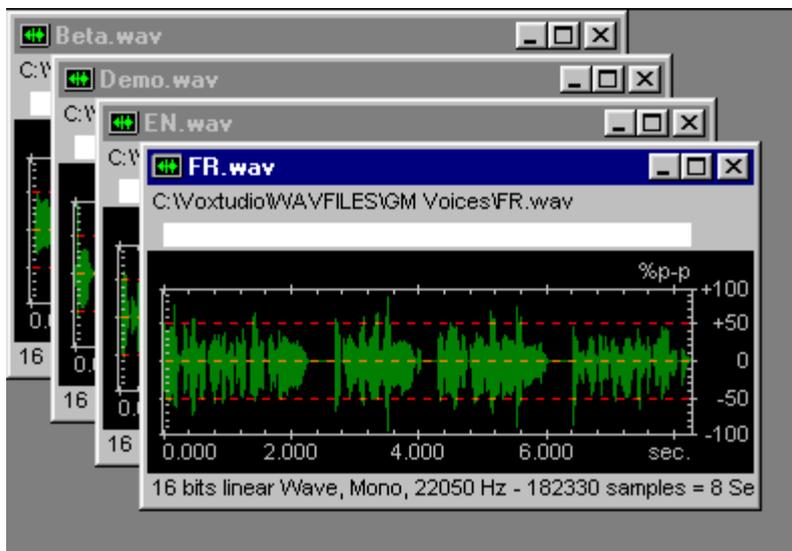
real 16-bit sound card. You can also play stereo files on mono systems. Vox Studio can play and accept stereo input files but, of course, always generates mono files for telephony purposes. The Sound Mapper also sometimes allows using sampling frequencies normally not offered by your sound device. You may then be able to work at sample rates that your card's hardware/driver cannot support directly. To see if you have the Sound Mapper installed simply click on the drop down listbox in the "Device" frame, all available devices will show.

5.8. Window menu

If you have lots of sound files open simultaneously in Vox Studio it will, from time to time, be necessary to reorganize the way files are shown by repositioning the graph windows. Rather than do it all by hand, window-by-window, you can use the Cascade and Tile commands to reposition all windows automatically.

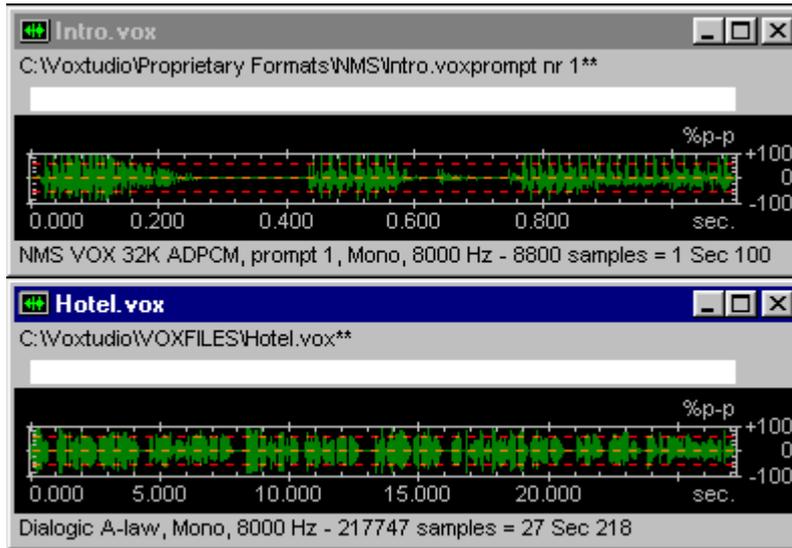
[Cascade](#)
[Tile Horizontally](#)
[Tile Vertically](#)

5.8.1. Cascade



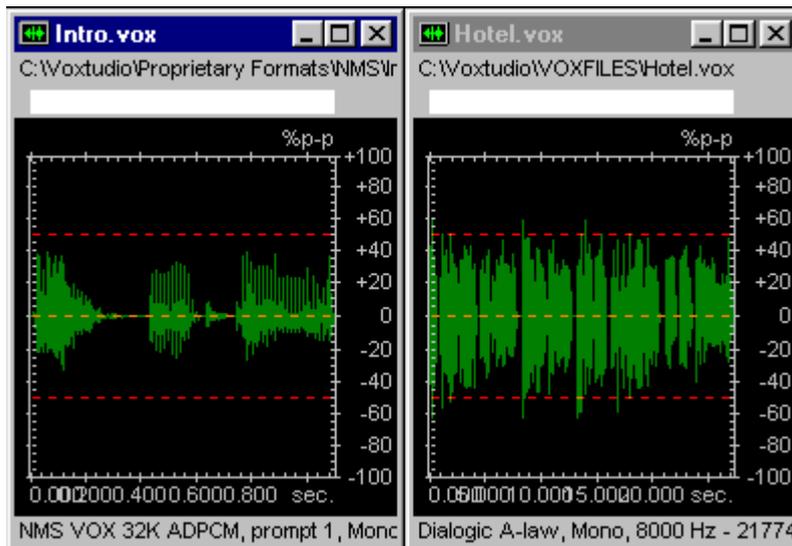
After a "cascade" command all windows will be positioned in the graph pane with the active window fully displayed on top and all other windows slightly offset from and positioned under the topmost window. Only the topmost window will be fully visible.

5.8.2. Tile Horizontally



After a "tile horizontally" command all windows will be positioned neatly one above the other in the graph pane so that all of them will be fully visible. As a result tiled windows will typically be smaller than cascaded windows (where only the topmost window is fully visible); this is especially true if you have many windows.

5.8.3. Tile Vertically



After a "tile vertically" command all windows will be positioned neatly next to one another in the graph pane so that all of them will be fully visible. As a result tiled windows will typically be smaller than cascaded windows (where only the topmost window is fully visible); this is especially true if you have many windows.

5.9. Help menu

This menu provides on-line help about Vox Studio. It is also here that you can look up how to obtain direct support, web support, or how to register your copy of Vox Studio.

[Contents](#)

[Search](#)

[License Registration](#)

[Print the license](#)

[Support](#)

[Xentec on the web](#)

[About Vox Studio](#)

5.9.1. Contents

Opens the Vox Studio help file at the very first topic.

From there you can navigate using hypertext jumps, browse with the left and right arrow buttons in the help tool bar or search for a particular help topic.

You will find hypertext jumps to help topics as well as a glossary of voice processing (computer telephony) terms at the back of the documentation.

5.9.2. Search

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. This command opens the help file and activates the "Search" feature.

5.9.3. License Registration

The first time Vox Studio is run it will open this type of dialog box:

Vox Studio registration

Please fill-in all identification fields below

Name:

Company:

Street:

City:

State:

Zip code: Country:

Phone: Fax:

To register, fill-in the License ID and associated License Key below. Registered copies of Vox Studio are linked to a License ID which identifies you uniquely. Every License ID is tracked.

You are therefore responsible for ensuring correct usage of this program, according to the terms of the license agreement. Licenses and Keys cannot be returned and you remain responsible for their correct usage. In case of doubt click <Cancel> now.

License ID:

License Key:

Please enter the complete information in all fields. If you are using Vox Studio in demo mode, and have not purchased a license, leave the License ID and License Key fields as they are and Vox Studio will run in evaluation mode. If you purchased a license enter your personal licensing information in those fields. Please make certain that you copy the personal ID and Key information exactly. Respect capitalization and make sure you do not substitute O (upper case oh) for 0 (zero) or 1 (one) and I (upper case i) for l (lower case L). If you can, copy and paste this licensing information into the fields from a pure text file. Do not paste this from a word processor, as you are likely to introduce invisible formatting characters in doing so.

5.9.4. Print the license

Prints a dated registration card you can use whenever you need to contact us, for support or upgrades. Fax this document with your request. If you purchased from us directly, fax it or mail it to Xentec at +32 2 757-0777. If not, send it to your local reseller.

5.9.5. Support

- The Support command pops up a box that gives you the coordinates of Xentec, the company which developed Vox Studio, and those of your local reseller (if applicable).

- This is where you will find address, phone, fax and email information to contact Xentec, or your direct reseller, for new orders, technical support, suggestions for new features, complaints or compliments. We would love to hear from you in order to improve Vox Studio.
- **Important:** to request pricing, place new or upgrade orders, and obtain technical support, always contact the company from whom you originally purchased this product.

5.9.6. About Vox Studio

The About Vox Studio command pops up a box that gives you program version information and serialization information identifying the legal owner of this package.

In order to protect you, the legal owner of this package, and trace any abusive copying of the program, your serialization information is encoded in various Vox Studio program files and is fully traceable.

5.10. Toolbars, buttons, keyboard and mouse

This chapter describes the toolbar button shortcuts which provide access to menu commands that otherwise would require a few more mouse clicks to access. Also described in this chapter are the contextual menus that are accessible by a mouse button right-click on the graph windows.

[Toolbar buttons](#)

[Player/Recorder buttons](#)

[Right-click contextual menu](#)

[Keyboard shortcuts](#)

5.10.1. Toolbar buttons

The tool bar, located just below the menu bar, is a shortcut for the most-used commands.



The icons from left to right and top to bottom represent the following commands:

- File/Exit 
- File/New 
- File/Open 
- File/Close 
- File/Close All 

- File/Save 
- File/Save All 
- File/Properties 
- View/Scroll Left 
- View/Scroll 
- View/Scroll Right 
- View/Zoom In 
- View/Zoom All 
- View/Zoom Out 
- View/Previous View 
- View/Next View 
-
- Edit/Undo 
- Edit/Cut 
- Edit/Copy 
- Edit/Paste 
- Edit/Delete 
-
- Convert/Transform Active 
- Convert/Convert Active 
- Convert/Batch Convert 
- Prompters/Prompter 
- Prompters/Tape Loader 
- Prompters/File Slicer 

5.10.2. Player/Recorder Buttons

The Player/Recorder buttons are located at the bottom of the graph pane and look like this:



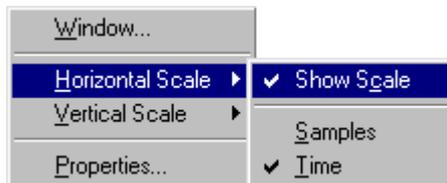
The buttons from left to right represent the following commands:

- Monitor Sound Input 
- Record 
- Fast Rewind 
- Play  (Play All or Play Window or Play Selection)
- Pause 
- Stop 
- Fast Forward 

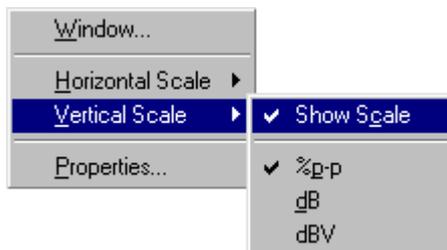
Play a List of Files  This is also the drag and drop play target (drop sound files on this button and they will play in Vox Studio)

5.10.3. Right-click contextual menu

When you right-click with the mouse on a graph window the following context-sensitive menu opens:



You can display or hide the horizontal tick marks and values and you can opt for a display of samples or time units.



You can display or hide the vertical tick marks and values and you can opt for a display in % peak-to-peak, dB and dBV (or dBm if dBm is what you selected in Tools/Defaults/dBV-dBm Calibration).

Selecting "Window" is the equivalent of the View/Window menu command.
Selecting "Properties" is the equivalent of the File/Properties command.

5.10.4. Keyboard shortcuts

- **Editing shortcuts:**

The editing shortcuts are the usual shortcuts found in most Windows programs:

Ctrl-x Cut selected sound to internal clipboard
 Ctrl-c Copy selected sound to internal clipboard
 Ctrl-v Paste internal clipboard into sound at cursor position (or at end of sound)
 Ctrl-z Undo the last operation
 Ctrl-a Selects all (the entire sound)
 Delete Deletes the selected sound

- **Player and Recorder shortcuts:**

p Starts playback
r Starts recording
s Stops playback or recording in recorder
a Pauses playback or recording in recorder
Space The Space Bar alternatively starts or stops playback in the main window.

- **Prompter and Tape Loader shortcuts:**

Space The Space Bar alternatively starts or stops recording in the Prompter and Tape Loader windows.

- **Other keyboard shortcuts:**

All menu commands and dialog box selections that have an underlined character in the command text can be controlled by issuing an "Alt-UnderlinedCharacter" keyboard command. For instance, to activate the Edit menu you would press Alt-e on the keyboard.

6. File Formats

This chapter describes the file formats for the script file used by the Prompter/Tape Loader/File Slicer/Group-Ungroup commands and also the telephony file formats currently supported in Vox Studio.

[Script File Format](#)
[Sound File Formats](#)

6.1. Script File Format

A script file is used by the prompter, by the tape loader and by the indexed file creation modules of Vox Studio. To make things simpler, the script file format is suitable for all these operations. The same script file can be used to record files with the prompter and to later group the files into multi-prompt indexed files. Also, the script format is such that any text editor is able to create a usable script file. A demo script file, prompts.txt, comes with your Vox Studio CD-Rom. There is a slight difference in the script file formats for Dialogic or for NMS indexed files. But the structure of the script files is similar enough that one can de-index one file format and use the same script file to re-index into the other file format.

Script files for Dialogic indexed files

A script file is a pure text file. All lines are terminated with the Enter key (Carriage Return / Line Feed pair). The file begins with "Begin Script" and ends with "End Script". The filenames have NO extension. The extension is ".wav" by default for the Prompter and Tape Loader. The annotation line is a short one-line description or title for each file. The text for the prompts uses as many lines as necessary and is followed by an additional empty line (a lone CR/LF pair). This text appears in the main window when you are using the Prompter. If annotations are not required by your system, these may be replaced by an empty line (a lone Enter or CR/LF pair), but they cannot be removed. The same is true for prompt text lines: if you do not use the prompt text (when using the Tape Loader for instance or when grouping files) you may replace it by an empty line if you prefer. You do not have to replace unused lines by blanks. If you leave them in, the same script file will be usable for the Prompter, the Tape Loader, the File Slicer and the Group operation. Use a text editor to generate or edit script files. Programs that come with Windows, like "Notepad" and "WordPad" (save as text), can produce prompt script files. There should be no additional blank lines (CR/LF pairs) to those shown below. Here is the layout for a Dialogic script file:

```
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 pair)
(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 pair)
(CR/LF pair)
and so on...
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 pair)
 (CR/LF pair)
 End Script (CR/LF pair)

Script files for NMS indexed files

A script file is a pure text file. All lines are terminated with the Enter key (Carriage Return / Line Feed pair). The file begins with "Begin Script" and ends with "End Script". The filenames have NO extension. The extension is ".wav" by default for the Prompter and Tape Loader. The prompt index is a sequence number for each prompt and is required by NMS applications. The text for the prompts uses as many lines as necessary and is followed by an additional empty line (a lone CR/LF pair). This text appears in the main window when you are using the Prompter. If you do not use the prompt text lines (when using the Tape Loader for instance or when grouping files) you may replace them by an empty line if you prefer. You do not have to replace unused lines by blanks. If you leave them in, the same script file will be usable for the Prompter, the Tape Loader, the File Slicer and the Group operation. Use a text editor to generate or edit script files. Programs that come with Windows, like "Notepad" and "WordPad" (save as text), can produce prompt script files. There should be no additional blank lines (CR/LF pairs) other than what is shown below. Here is the layout for a NMS script file:

Begin Script (CR/LF pair)
 (CR/LF pair)
 Filename for prompt 1 (no extension here) (CR/LF pair)
 Prompt index for prompt 1 (CR/LF pair)
 Text of prompt 1 on as many lines as needed (CR/LF pair)
 (CR/LF pair)
 Filename for prompt 2 (no extension here) (CR/LF pair)
 Prompt index for prompt 2 (CR/LF pair)
 Text of prompt 2 on as many lines as needed (CR/LF pair)
 (CR/LF pair)
 and so on...
 Filename for prompt n (CR/LF pair)
 Prompt index for prompt n (CR/LF pair)
 Text of prompt n on as many lines as needed (CR/LF pair)
 (CR/LF pair)
 End Script (CR/LF pair)

6.2. Sound File Formats

Essentially, Vox Studio knows two generic groups of sound file formats: the multimedia formats (usually known as ".wav" files) and the telephony formats (often known as .vox files, but it can be anything you want).

[Script file format](#)
[Bicom format](#)
[Centigram formats](#)
[Dialogic formats](#)
[Elan Informatique format](#)
[Group 2000 formats](#)
[IBM Directtalk format](#)
[InterVoice formats](#)

[ITU-CCITT formats](#)
[Microlog Intela formats](#)
[Natural Microsystems \(NMS\) formats](#)
[NewVoice formats](#)
[Next and Sun formats](#)
[Nortel Generations formats](#)
[OKI file formats](#)
[Philips VoiceManager formats](#)
[PhoneBlaster formats](#)
[Raw PCM formats](#)
[Rockwell formats](#)
[SCII format](#)
[SoundDesigner II format \(Mac\)](#)
[Voicetek Generations formats](#)
[Windows Wave formats](#)

6.2.1. Bicom format

The proprietary Bicom format supported in Vox Studio is of the 4-bit ADPCM type. The supported sample rate is 8KHz. This is a headerless file format.

6.2.2. ITU (CCITT) formats

The ITU (CCITT) coding formats Vox Studio supports are all sampled at 8 KHz. Refer to the ITU G-series publications for a detailed description of the supported algorithms.

Vox Studio supports two ITU (CCITT) companding algorithms (G.711) and 8 ITU (CCITT) ADPCM compression algorithms (G.721, G.726). The ITU ADPCM algorithms are very processor intensive (you need a fast PC).

G.711

Vox Studio supports the ITU G.711 specification for A-law and Mu-law companding. See the glossary in this documentation for a brief description of A-law and Mu-law.

G.721

32 KBps ADPCM, 4 bits at 8,000 samples/s. See the glossary in this documentation for a brief description of ADPCM.

G.726

40 KBps ADPCM, 5 bits at 8,000 samples/s
 32 KBps ADPCM, 4 bits at 8,000 samples/s
 24 KBps ADPCM, 3 bits at 8,000 samples/s
 16 KBps ADPCM, 2 bits at 8,000 samples/s

6.2.3. Centigram formats

Three Centigram ADPCM file formats are supported under Vox Studio. All are ADPCM variants sampled at 8 KHz.

The supported Centigram formats are:

- 32K ADPCM
- 24K ADPCM
- 16K ADPCM

These formats have a file header that contains all the necessary information for decompression. Therefore, when Vox Studio opens such a file, it will not ask you to enter additional information such as sample rate or coding algorithm.

6.2.4. Dialogic formats

There are several Dialogic telephony-type file formats as enumerated below. The usual file extension is ".vox", but this a habit, not a rule. Some Dialogic-based voice processing system developers use different file extensions.

Dialogic telephony cards use sampling rates of 6.0, 6.053, 8.0, 8.117 or 11.025 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 requires.

One of the annoying characteristics of native Dialogic telephony file formats is that they contain only raw data (except for the new Dialogic ".wav" formats). Most 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 behavior regarding files with or without headers. When Vox Studio reads ".wav" files there is no need to tell it what the file contains; this information is found in the file header itself. When Vox Studio reads native Dialogic ".vox" 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, with or without a header, 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 want 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 a number of ".vox" files.

Here are the Dialogic telephony sound file formats known to Vox Studio today:

ADPCM (OKI variant) and ADPCM Wav

ADPCM stands for Adaptive Differential Pulse Code Modulation. There are various flavors of ADPCM. The algorithm we have implemented in this version is the original algorithm used by Dialogic voice processing hardware. Future versions of Vox Studio will support more flavors of ADPCM required for other telephony hardware.

OKI ADPCM, as used by Dialogic, 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 represents 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 signal errors and the sound quality is slightly affected, but it remains sufficient for many telephony applications. Naturally, 8 KHz ADPCM sounds MUCH better than 6 KHz ADPCM.

Traditionally, 6 KHz ADPCM is also called 24 Kbps (6KHz x 4 bits) and 8 KHz ADPCM is called 32 Kbps (8KHz x 4 bits). This is a very confusing way of defining the sound coding algorithm used as, for instance, some other ADPCM algorithms produce 24 Kbps which is in fact 3-bit data sampled at 8 KHz!

Not many people know that some cards use 6.0 and 8.0 KHz sampling rates and other (very old cards) 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, the chances are nobody will ever notice the slight difference in pitch. However, if the files contain frequency-sensitive material, say DTMF data streams, then the 1.5% difference may in fact cause very severe problems.

Vox Studio has the capability to convert to and from, indexed ADPCM files (".vap" files) as well. These are files that contain more than one voice message per physical file, with a header (at the beginning of the file) that contains pointers to the start of each separate voice message. This technique was introduced mainly to circumvent the problems good old DOS had when too many files were opened simultaneously by a running application.

The Dialogic ADPCM ".wav" format uses the same coding as normal Dialogic files (it can contain sound coded in A-law, Mu-law or ADPCM) but it has a RIFF-standard file header instead of just raw data. One more sample frequency is provided: 11.025 KHz.

A-law and A-law Wav

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 (Telephone Companies) throughout Europe. In the USA a similar algorithm called Mu-law is used.

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 analog 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.

Although the "normal" sample rate for A-law telephony is 8 KHz, most Dialogic cards allow using A-law at 6KHz as well, and Vox Studio supports this.

The Dialogic A-law ".wav" format uses the same coding as above but has a RIFF-standard file header instead of just raw data. One more sample frequency is provided: 11.025 KHz.

Mu-law and Mu-law Wav

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. In Europe a similar algorithm called A-law is used.

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 analog 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.

Although the "normal" sample rate for Mu-law telephony is 8 KHz, most Dialogic cards allow using Mu-law at 6KHz as well, and Vox Studio supports this.

The Dialogic Mu-law ".wav" format uses the same coding as above but has a RIFF-standard file header instead of just raw data. One more sample frequency is provided: 11.025 KHz.

6.2.5. Elan Informatique format

Elan Informatique is a French manufacturer of telephony cards and interfaces. They have a lot of experience in applying the CNET text-to-speech (PSOLA) technology in telephony applications.

The Elan Informatique coding format is a variant of G.721 CCITT 32 Kbps ADPCM.

6.2.6. Group 2000 formats

The Group 2000 file formats supported by Vox Studio are formats with a file header. The usual extension for Group 2000 formats is .vsn

You can select between the following codings:

OKI ADPCM at 6 or 8 KHz

A-law PCM

Mu-law PCM

6.2.7. IBM Directalk formats

Vox Studio currently supports the IBM DirecTalk A-law format. Support for the other formats is imminent.

6.2.8. IMA ADPCM

ADPCM stands for Adaptive Differential Pulse Code Modulation. ADPCM is a lossy compression mechanism. There are various flavors of ADPCM. This particular algorithm was suggested by the International Multimedia Association (IMA). IMA ADPCM compresses data recorded at various sampling rates. Sound is encoded as a succession of 4-bit or 3-bit data packets. Each data packet represents the difference between the current sampled signal value and the previous value. The compression ratio obtained is relatively modest. As an example: 16-bit data samples encoded as 4-bit differences result in 4:1 compression format.

IMA ADPCM is similar to Intel's DVI audio format.

IMA ADPCM is directly supported on most Windows implementations as a native format. Although the quality of IMA ADPCM voice files is not great, the files are portable. There is a real advantage in having compact files that can be played on most Windows PCs.

Windows supports IMA ADPCM ".wav" files in 4-bit format only and at sample rates between 8 and 41KHz.

Vox Studio supports IMA ADPCM ".wav" files in 4-bit or 3-bit format at sample rates from 6KHz to 64KHz.

6.2.9. InterVoice formats

Vox Studio supports the InterVoice 64K, 32K, 24K, and 16K A-law and Mu-law ADPCM file formats.

6.2.10. Microlog Intela formats

The Microlog Intela file formats supported by Vox Studio are formats with a file header. The usual extension for Microlog Intela formats is ".vsn".

You can select from the following coding types:

OKI ADPCM at 6 or 8 KHz

A-law PCM

Mu-law PCM

6.2.11. Microsoft ADPCM

ADPCM stands for Adaptive Differential Pulse Code Modulation. ADPCM is a lossy compression mechanism. There are various flavors of ADPCM. This particular algorithm was suggested by Microsoft; its quality is similar to IMA ADPCM. MS ADPCM compresses data recorded at various sampling rates. Sound is encoded as a succession of 4-bit nibbles. Each nibble represents the difference between the current sampled signal value and the previous value. The compression ratio obtained is relatively modest: 16-bit data samples encoded as 4-bit differences result in 4:1 compression format.

Microsoft ADPCM is directly supported on most Windows implementations as a native format. Although the quality of IMA ADPCM voice files is not great, the files are portable. There is a real advantage in having compact files that can be played on most Windows PCs.

Windows supports Microsoft ADPCM ".wav" files at sample rates between 8 and 41KHz.

Vox Studio supports IMA ADPCM ".wav" files at sample rates from 6KHz to 64KHz.

6.2.12. Natural Microsystems formats

Vox Studio supports 2 companding and 3 compression formats used by NMS (Natural MicroSystems) cards (see below). Stand-alone NMS files usually carry the ".vce" extension while indexed files usually

carry the ".vox" extension. Vox files usually (but not always) contain multiple prompts. NMS vox files that contain only a single prompt are called single-prompt vox files in Vox Studio. Both ".vce" and ".vox" files have a file header. Vox Studio converts to and from NMS ".vce" files or to and from NMS single-prompt ".vox" files. Vox Studio has a separate command to group prompt files into indexed ".vox" files or ungroup ".vox" indexed files to stand alone files.

To summarize: NMS ".vce" files are non-indexed files and contain a single voice prompt. NMS ".vox" files are indexed files and can contain either multiple voice prompts or a single prompt. Vox Studio converts directly from other single-prompt formats (".wav" for instance) to ".vce" files or to "single-prompt vox" files. To produce "multi-prompt vox" files you have to use the "Group NMS" command that does on-the-fly conversion to NMS formats. It is thus possible to group ".wav" files, for instance, into a vox file; conversion is done automatically by Vox Studio.

Companding laws

Vox Studio supports the NMS file format with both A-law and Mu-law companding at 8,000 samples per second.

NMS ADPCM

Vox Studio supports the NMS file formats with ADPCM compression at 32 Kbps, 24 Kbps and 16 Kbps, all at 8,000 samples per second. 32 Kbps is by far the most popular coding scheme for NMS cards.

6.2.13. Newvoice formats

The NewVoice CVSD format is supported at 24 Kbps, 32Kbps, 64Kbps.

6.2.14. Next and Sun formats

Vox Studio supports the following NeXT / Sun formats:

- 8 bits linear at 8KHz, 22.05 KHz, 44.1 KHz
- 16 bits linear at 8KHz, 22.05 KHz, 44.1 KHz
- A-law at 8KHz, 22.05 KHz, 44.1 KHz
- Mu-law at 8KHz, 22.05 KHz, 44.1 KHz

6.2.15. Nortel Generations formats

The Nortel Generations file formats supported by Vox Studio are formats with a file header. The usual extension for Nortel Generations formats is ".vsn".

You can select from the following coding types:

- A-law PCM
- Mu-law PCM
- OKI ADPCM at 6 or 8 KHz

6.2.16. OKI file formats

Vox Studio supports 4-bit OKI ADPCM in raw and ".wav" file formats with sample rates at 8 and 16 KHz.

6.2.17. Philips VoiceManager formats

The Philips file formats supported by Vox Studio are formats with a file header. The usual extension for Philips VoiceManager formats is ".vsn".

You can select between the following codings:

A-law PCM

Mu-law PCM

OKI ADPCM at 6 or 8 KHz

6.2.18. PhoneBlaster formats

PhoneBlaster files are files that have a file header. The ones supported by Vox Studio correspond to the format used by the SuperVoice application from Pacific Image. That application comes with the PhoneBlaster card.

The PhoneBlaster SuperVoice coding algorithm supported by Vox Studio is:
4 bits ADPCM at 7.2 KHz

6.2.19. Raw PCM formats

The "Raw" PCM formats are generic formats not readily associated with a card or system manufacturer. Under "Raw" formats you will find only headerless formats, sometimes at unusual sampling rates. For instance, A-law is a companding law that is always used at 8,000 samples per second in the telephony world. If you select "Raw PCM Formats" we will in fact allow you to produce A-law files at any frequency from 6 KHz to 64 KHz; the same is true for the other "raw" formats. The "raw" formats are provided for knowledgeable users; use them with circumspection.

16-bit and 8-bit Linear PCM

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 from 6,000 up to 48,000 samples per second. Of course, this is overkill for most standard telephony applications.

Vox Studio uses a high-resolution linear representation of signals internally to ensure the 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 that are utilized, the more accurate the signal representation will be. 8-bit PCM represents signals digitized into 256 discrete levels. 16-bit PCM represents signals digitized into 65,536 discrete levels. The higher the resolution, the more hi-fi your reproduced sound gets. Also, the more samples that are taken per second, the better the reproduced sound gets. Obviously, 16-bit linear PCM sampled at 48 KHz represents a data stream of 768,600 bps, 32 times more than 6 KHz ADPCM at 4 bits! Be careful when you select extreme resolutions, you may fill your hard disk much faster than you expect.

We strongly advise against the use of 8-bit resolution master recordings, even for telephony, unless absolutely required. Use 16-bit resolution wherever possible.

6.2.20. Rockwell formats

The Rockwell formats supported by Vox Studio are the ADPCM coding formats understood by the voice-enabled Rockwell modem chips. Many voice/data/fax modems on the market use the Rockwell chip-set. If your application uses one of those, this may be the format you need.

Vox Studio supports Rockwell's 4 bits ADPCM, 3 bits ADPCM and 2 bits ADPCM, all sampled at 7,200 samples/s.

When specific boards use Rockwell chips but have defined their own file formats these are treated as distinct formats by Vox Studio; for instance the Creative Labs PhoneBlaster format, which is a file format that includes a header.

6.2.21. SCII format

SCII is a French manufacturer of telephony cards and ISDN interfaces. The SCII coding format is a variant of G.723 CCITT 32 KBps ADPCM.

6.2.22. SoundDesigner II format

The ProTools SoundDesigner II format comes to us from the Mac world. It is used frequently in studio environments. Vox Studio supports the 16-bit SoundDesigner format at 44.1 and 48.0 KHz. Note that these are also the sample rates used on CDs and DATs.

6.2.23. Voicetek (Aspect) Generations formats

The Aspect (previously Voicetek) file formats supported by Vox Studio are formats with a file header. The usual extension for Aspect (Voicetek) formats is ".vsn".

You can select from the following coding types:

32 KBps OKI ADPCM at 8 KHz
 24 KBps OKI ADPCM at 6 KHz
 A-law PCM
 Mu-law PCM

6.2.24. Windows .wav formats

A large number of file coding types for ".wav" files have been registered with Microsoft and the following are supported for ".wav" files in Vox Studio: 8-bits linear PCM, 16-bits linear PCM, A-law, Mu-law, Dialogic ADPCM, OKI ADPCM.

The industry standard format for multimedia sound files is the ".wav" PCM format. These files are usually (but not necessarily) recorded at 11.025 KHz, 22.05 KHz, 44.1 KHz or 48 KHz sampling rates with a resolution of either 8 or 16 bits.

In addition to 11, 22, 44 and 48 KHz files Vox Studio can also record and play back ".wav" files at 6.0, 6.053, 7.2, 8.0, 8.117, 16, 24, 32 and 64 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 gets that information from the ".wav" files themselves (this is not so with telephony type files which usually contain raw data only).

Vox Studio can also write the very old ".wav" file format used by some obsolete ".wav" utilities. There is an option in the output file format section of the Defaults menu that allows you to write the old-format files. Do not use this unless you absolutely must do so.

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
- 7,200 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)
- 48,000 Hz CD quality (if 16 bits)

7. Tips and Techniques

This chapter gives useful recording and conversion tips. Please read this chapter before spending a lot of money on equipment.

[Sound card quality](#)

[Mixer, volume and editor applets](#)

[Clean recordings to start with](#)

[Conversion and sound quality](#)

[Sampling frequencies of older cards](#)

[My files do not sound right](#)

[Sample file](#)

7.1. Sound card quality

Although sound card bargains certainly exist, our experience shows that the sound cards that consistently give good or excellent results are not the rock-bottom-priced, unlabeled types. 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.

You need to be warned: there are substantial differences in quality between the various sound cards available on the market today. We have yet to see a very cheap card that sounds great and has low noise but we certainly have seen some very expensive junk. The most important aspects are:

- Some multimedia sound cards work very well but only when recording or playing back at the standard multimedia 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 extremely important. If you hear superimposed 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 and some famous products don't.
- A sound card is not a must. You can play files back over a voice processing telephony card using your standard telephony software with your own voice processing application.
- Select a card that can do 16-bit recording and playback, and do it well from 6 to 64 KHz. The 8-bit cards, although usable, naturally introduce more quantization noise and are often of lesser quality. They will never give you good results and should be avoided. Of course, we are talking about the resolution of the recordings here, not about the complexity of the wave tables used to generate synthetic music.
- 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! This is very simple to test with Vox Studio. Mute your microphone input and start recording from that input. Then watch the recorded "silence" in Vox Studio in dB mode (right-click on the waveform and select Vertical Scale/dB). If you see anything at all above -90dB, your card does not produce pure silence (it probably does not).
- Some sound cards give better results (less noise) with microphones when the microphone is connected to the line input rather than the microphone input. Strange, but true. It is worth the

experiment.

- Make sure that your sound card records and plays back at the precise sampling rate you ask it to. We have encountered a sound card that was 9% off its 6,000 Hz sampling rate, in other words it was unusable. Some manufacturers cut corners and use simplified clocks. As a result the sample rate is sometimes not very accurate outside of the traditional 11, 22 and 44 KHz ranges. This is not audible if you record and play on the same card. This becomes very audible if you export the resulting sound file to another system. This is catastrophic if your files contain calibrated frequencies like the DTMF tones. As a test, record any input at 8 KHz for exactly one minute (use a stopwatch) then look at the recorded signal in Vox Studio. The bottom part of the window should indicate exactly 480,000 samples in the file.
- 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. It would be nice to be able to test it on Vox Studio and return it if it is not good enough for your telephony needs.
- If you are using a cheap sound card and find it to be of impeccable quality, let us know.

Good multimedia cards are sold with decent accompanying software. Look for a quality mixer (level adjustment) applet; you need one to adjust the input levels under Windows. Vox Studio does not provide such a tool; it comes standard with sound cards.

7.2. Mixer editor and volume applets

Vox Studio does not provide a mixer (sound level adjustor) for your multimedia sound card. These software applets always come with the sound card itself and they get installed in Windows when you install the drivers for your card.

When installed they typically look like this in the System Tray on the Taskbar: 

Mixers and volume control applets are often card-hardware-specific. The philosophy behind Vox Studio is to be card-independent. Use the specialized mixer applets that come with your card.

7.3. Clean recordings to start

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. Many ingredients influence the quality of a recording: the microphone, the room's acoustical characteristics, and the position of the microphone with respect to the room and the speaker all play a very important role.

Use a hyper-directional microphone to avoid recording surrounding noises (PC fan noise for instance) and room echo. Do this even if you have a silent studio area. Better microphones will also have a lower inherent (electrical) noise factor. Cheap microphones are usually not directional and pick up all sorts of noises and room echoes, giving the recording a very amateurish sound.

Use a low-impedance hyper-directional microphone and keep mic cables short, shielded and away

from power cables and outlets. Ground your PC system at one point only. Hum (at 50 or 60 Hz) in a lab or office environment is an interference that plagues many aspiring "sound engineers". If possible, do not record frequency content below 100Hz; it does not get transmitted over the telephone network bandwidth anyway.

Avoid recording in concert halls, empty basements or stuffy closets. The room's reverberation time and echoes have a very palpable influence on sound "quality". For simplicity, imagine reverberation being the time it takes a hard clap with the hands to die out (decay) until you can't hear it anymore, while echo is the fact that you can actually hear a second or maybe even a third clap. The reverberation time depends on the structure of the walls, floor and ceiling. The more they absorb sound, the shorter the reverberation time will be. The ideal situation is when the reverberation time is short and independent of the sound's frequency. This is very difficult to achieve in a normal office environment (but exists in professional studios), so you will have to compromise. Avoid large rooms without furniture and with glass windows (glass absorbs low frequencies and reflects high frequencies). If you have glass windows, use curtains to absorb reflections. Avoid unfurnished rooms with wall-to-wall carpeting and drapes over the walls (these only absorb the higher frequencies). Echo is something that is very disturbing and should be avoided by all means. The larger the room, the larger the echo risk. In the selection process for a studio room it is a very good idea to clap loudly with your hands and listen for sound decay and possible echoes. Move around in the room and clap your hands to determine how this changes the room response and where the best location is. Your colleagues may think you have gone nuts, but ignore them. This may seem strange but square or rectangular rooms with flat ceilings are the worst because standing acoustic waves are created in all three dimensions; asymmetrical rooms with sloping walls and furniture are better because standing waves are less likely to be created. In conclusion: you are unlikely to find a perfect studio room in your office, so establish your recording microphone in a well-furnished, preferably asymmetrical, mid-sized room where a clap with the hands decays in less than half a second, use a directional microphone and forget about the rest.

The spatial impression and richness of a recording depend on the location of both the microphone and the speaker in the recording room. The distance between the speaker and the microphone influences the spatial impression because you change the ratio between direct sound and reverberated sound. The closer the speaker is to the microphone the closer he will sound but his voice will also seem warmer because low frequency content will increase. The further the speaker is from the microphone the more spacy and distant he will sound and the more you will hear room echoes. The above is also a function of the directionality of the microphone, therefore trial and error recordings are needed to determine the optimal position of the speaker with respect to the microphone. For speech, the speaker at arm's length distance from the microphone is usually about right for a neutral impression. In any event, keep the microphone far enough from the speaker's lips to minimize "plops" and breathing sounds. If your microphone rests on a table, try and minimize the interference between direct sound and sound reflected by the table surface by placing the microphone as close to the table surface as possible. Another way of avoiding this is to use a highly directional microphone (again).

Another, very important factor is the quality of the voice itself. Use experienced speakers with a clear pronunciation and telephone-friendly voice. A voice that sounds great on radio or TV may not sound all that great over the telephone system. Perform preliminary tests. Some speakers have a propensity to produce DTMF sounds; avoid their services. Other speakers pronounce the letter S as if they were sssnakes trying to hypnotize a prey--not good either. Test the speakers' voice quality before and after conversion to the desired telephony format and play a few test prompts over the telephone network via a voice telephony card.

Use a PC with no fan (e.g. a portable PC) or one with a very quiet 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.

Avoid recording with an 8-bit sound card or with a 16-bit sound card set for 8-bit resolution. We have

only included 8-bit capability in Vox Studio for those rare occasions where you absolutely need to use an existing 8-bit resolution file and do not care about quality. Otherwise, you should stick to 16-bit recordings.

If you hear hissing noises in your recordings that cannot be related to surrounding noise reaching the microphone, you are having electromagnetic interference problems. Swap your sound card for another, better one. Move the sound card to another free slot in your PC. 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.

Do not record in a room where others work. Close your window to keep outside noise away. Avoid recording under the flight path of Concorde. Record in the evenings, at night or during weekends if you cannot avoid hearing slight workday noises. Even slight noises get recorded and cause problems later.

You would be amazed at how many little noises actually disturb a place you think is quiet. The ticking of the clock on your office wall, the humming of your fluorescent lights, hot water flowing through your heating system, the air conditioning, a noisy hard disk, the cooling system of your PC. All of these add up. 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, because your microphone is likely to hear it too.

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? Look at the recorded signal in dB view. Is everything that is visible below -60dB or less? If not, solve this problem before you go one single step further. Clean recordings convert very nicely to other formats, recordings with background noises or hissing don't.

Record at reasonable levels. You must absolutely avoid compensating for low-level recordings by pumping the gain up at a later stage in the process! Avoid saturation while recording, except perhaps for very brief peaks.

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 mouse pad. Do not locate the keyboard on the desk that carries the microphone or do use a thick sound-absorbing mat. Avoid kicking the recording table. An office chair on wheels can generate noise when you move. Chairs with springs and floppy armrests can be pretty bad at making noises too. Test this.

One procedure to obtain good quality telephony prompts (there are many variations) 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. Avoid recording directly to 8-bit resolution files.

If you cannot follow the above guidelines, have a professional studio record the ".wav" files for you, or have them record a tape, DAT, mini-disc or Audio CD and use the Vox Studio Tape Loader to digitize their recording and convert the files to telephony formats and sample frequencies.

Always test the quality of your master files before proceeding to convert them.

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, normalization or other conversion manipulations.

7.4. Conversion and quality

Converting files from one format to another does not affect sound quality if all that changes is the data-recording format. Compression, companding and sample rate changes of digitized data are subject to the laws of physics and do cause a slight or sometimes drastic alteration in sound quality. First you have the expected degradation resulting from the target sample rate conversion itself. 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 have to accept them. The resulting quality is not much different than what the file would have been if it were recorded directly at the lower sample rate or resolution. For example, converting a hi-fi 16-bit linear ".wav" file at 44 KHz into a 4-bit ADPCM file at 6 KHz does cause a very noticeable (and unavoidable) degradation in sound quality. This is because a 44 KHz sampled signal can contain components up to a frequency of 22 KHz, while a 6 KHz signal can only contain components up to 3 KHz! In effect you are going from a hi-fi quality signal to a low telephone-quality signal. Also, only 4 bits now represent your original 16-bit signal. Nobody can avoid these basic laws of physics. This is not as dramatic as it sounds, because in the end you are going to use this message over telephone lines anyway and telephone bandwidth is only from 300 to 3,400 Hz. You also should expect a very slight but noticeable degradation resulting, essentially, from the resampling processes themselves. Record directly into the final target sample rate if your hardware allows it and does it well. 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, 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 of comparable quality. Always try to record your originals at 16-bit resolution, not at 8-bit.

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, however. You cannot gain bandwidth, and the signal information that was absent in the original will also be absent in the up-converted file. You cannot change a telephone-quality file into a hi-fi-quality file, but you can do the opposite.

Vox Studio uses digital signal processing to perform the signal conversions. Each conversion involves many mathematical manipulations of the recorded files. It is best to avoid unnecessary conversion steps and record or convert your master file directly to the desired target format if your sound card records well at that sample frequency. Many sound cards unfortunately do not have good recording capabilities below 11 KHz sample rates; they were optimized to be used at 11, 22 and 44 KHz and nothing else.

If you plan to edit your files, do this editing on a copy of your master ".wav" recording, before conversion to telephony format. Convert to telephony format as the last step in the process.

All the above may seem complicated but we only discussed it so you would understand the various parameters that can affect the quality of a recorded or converted message. The reality is that, in general, you can obtain excellent telephony quality prompts if you follow a few simple guidelines:

- Record the originals as 16-bit ".wav" files at 22 KHz or higher.
- Make sure your original recordings are spotless: good intelligibility, good sonority, no background noise, good volume, etc. If they are not, start again until they are. Do not attempt to convert unless the original recordings are perfect and sound professional.

Use Vox Studio to convert to the highest sample rate allowed by your telephony hardware (for instance, 8 KHz sample rates are better than 6 KHz) and use A-law or Mu-law rather than ADPCM coding if you can.

7.5. Sampling frequencies of old cards

Some very old Dialogic telephony cards (the ancient D/41-B for instance) use 6,053 Hz and 8,117 Hz sample frequencies. Newer cards use exactly 6,000 Hz and 8,000 Hz. Vox Studio can accommodate both types. You cannot buy the old cards anymore but they still are installed in systems in the field, so you need to know about this requirement.

The difference is not noticeable when playing back speech files from one type of card to another. However, when precisely pitched tones (DTMF for instance) are recorded, the difference may be significant. Check with your telephony card vendor for the exact sample rate of the card(s) you are using. Some telephony cards can play back at several programmable rates. In that case, check with your software vendor to select the exact sampling rate and file format that your applications need.

Remember that some telephony applications and cards need files sampled at 6 KHz, others at 8 KHz and even others at 7.2 KHz! Check with both your card vendor and the vendor of your application software platform.

7.6. My files do not sound right

When a user reports that his files do not sound right in his application he usually experiences one of the following phenomena:

- The file plays in the telephony system and has recognizable human speech but plays too fast and at too high a pitch. This means the correct coding algorithm was selected to play the file but the wrong sample frequency. Select a lower sample rate when playing back on the telephony system or, if that is not possible, a higher sample rate when recording with Vox Studio.
- The file plays in the telephony system and has recognizable human speech but plays too slow and at too low a pitch. This means the correct coding algorithm was selected to play the file but the wrong sample frequency. Select a higher sample rate when playing back on the telephony system or, if that is not possible, a lower sample rate when recording with Vox Studio. In some rare instances we have found that the hardware was in fact the culprit. We have seen a sound card that had a playback frequency 9% off the one we had programmed, at low frequencies. As a result files recorded at a higher sample rate, and then converted, sounded off-pitch when played back at 6 or at 7.2 KHz.
- The converted voice files sound "muffled" and seem to lack clarity. This is usually the result of a drastic reduction in sample rate that results in a perceived increase in low-pitched sounds. This is most audible if the voice talent has a voice with high bass content to begin with. Use the Intelligibility Filter option in the Vox Studio conversion dialog boxes to correct this effect.
- The file is playing but produces a very loud screeching sound. This means one has selected the wrong coding algorithm to record the file. If you chose an ADPCM variant try A-law or Mu-law for instance, or another form of ADPCM. The most efficient approach, of course, is to ask the

manufacturer exactly what coding and sample rate are required.

- The file is playing right, it is understandable and at the right speed but there is a short noise burst at the beginning of every file. This means that both the correct coding algorithm and sample rate were used but the wrong file format. One is probably playing a file in one of the file formats that have a file header at the beginning while the application expects a headerless file.

If you are unsure about the file format a particular platform needs, it may be useful to take a sample file (known to play correctly on the platform) and to bring it over to the PC where Vox Studio is installed. Then it is usually easy to identify the format of that file by playing it using Vox Studio. Use the same format for making new files.

7.7. Sample files

The sample file "test.wav" that comes with your distribution CD is a good vehicle to test the conversion, trimming, normalizing, and other functionalities available under this version of Vox Studio.

The current sample file is provided courtesy of recording studio:

Vert Foncé
16 rue des amandiers
37000 Tours
France
Tel: +33 (2) 4761-4205
Fax: +33 (2) 4761-5590
E-mail: verforce@club-internet.fr

Other quality sample files, from other studios, are provided on the distribution CD. These are not included in the downloadable package in order to limit the size of the downloadable file. These files are provided courtesy of their respective authors for testing and evaluation purposes only and should not be used in commercial applications. Contact information is provided on the CD. Contact the authors for additional information and for custom production requests.

8. Support section

You are entitled to technical support for your registered licensed copy of Vox Studio. Read the following paragraphs to understand how you can obtain support for Vox Studio.

[Registration](#)

[Support](#)

[Functional demo](#)

[Third party trademarks](#)

[Visit the Xentec web site](#)

[Send an e-mail to Xentec support](#)

[End-User License Agreement](#)

8.1. Registration

Vox Studio registration

Please fill-in all identification fields below

Name:	<input style="width: 95%;" type="text" value="John Doe"/>		
Company:	<input style="width: 95%;" type="text" value="Xentec NV"/>		
Street:	<input style="width: 95%;" type="text" value="De Helftwinning 2"/>		
City:	<input style="width: 95%;" type="text" value="Kortenberg"/>		
State:	<input style="width: 95%;" type="text" value="Brabant"/>		
Zip code:	<input style="width: 60%;" type="text" value="3070"/>	Country:	<input style="width: 40%;" type="text" value="Belgium"/>
Phone:	<input style="width: 60%;" type="text" value="32 2 757-0666"/>	Fax:	<input style="width: 40%;" type="text" value="32 2 757-0777"/>

To register, fill-in the License ID and associated License Key below. Registered copies of Vox Studio are linked to a License ID which identifies you uniquely. Every License ID is tracked.

You are therefore responsible for ensuring correct usage of this program, according to the terms of the license agreement. Licenses and Keys cannot be returned and you remain responsible for their correct usage. In case of doubt click <Cancel> now.

License ID:	<input style="width: 95%;" type="text" value="JOHNDOE654321"/>		
License Key:	<input style="width: 95%;" type="text" value="*****"/>		

Registered users have purchased a license for Vox Studio and have received a unique license ID and license key. The license ID and key are confidential and personal. Users should always test the free demo version before purchasing a full license. Once purchased, a software license is nonreturnable. Do not lose your licensing information; it will not be replaced. Do not show your license information to anyone; it is personal.

Your license ID is required for every technical support or upgrade request.

Vox Studio requires user information to be entered before you use the program. This is required once only, and may have been done for you by your distributor.

Only one installed version (and one nonutilized and noninstalled backup copy) per paid Vox Studio license is allowed to exist at any time. Vox Studio can be reinstalled whenever you need to, but only on one single PC for each purchased license. We trace any instance of Vox Studio to protect our legitimate customers, our distributors, and ourselves. Violations of the software license will always be prosecuted to the maximum extent permissible under the law.

If you are upgrading Vox Studio to a newer version, the new version replaces the previous one. The previous version of Vox Studio covered by that license has to be taken out of circulation and discarded. Failure to do so is a violation of your Vox Studio license agreement.

When you stop using a license on a PC to install the license on another one, you should unregister the license from the old PC before installing it on another PC. A special "Unregister" button is provided to that effect. Unregistering simply returns the program back to demo mode.

From time to time, Xentec may decide to change license keys for any version of Vox Studio. In that case registered users can obtain a new key for their existing equivalent license free from Xentec.

Please read the End User License Agreement for full details on the Vox Studio software license. The End User License Agreement is a binding contract between Xentec and its customers.

8.2. Support

Contact your local distributor or Xentec for Vox Studio technical support, upgrades and sales information. The support rules are simple:

- Technical support is provided to you **by the company who sold you Vox Studio**
- Technical support is provided to the legitimate owner of a valid registered license



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Tel.: +32 2 757-0666
Fax: +32 2 757-0777
E-mail: support@xentec.be
Web: <http://www.xentec.be>

If you have problems using Vox Studio please start by looking for a possible solution to your problem

in the on-disk manual or visit our web site at <http://www.xentec.be> . Try to run Vox Studio when absolutely no other programs are running. Make sure you use the correct file format. If this fails, contact your local distributor or Xentec for support (our address is shown above). We will be happy to be of service. As a licensed customer you will get our undivided attention, and we will do our best to help you. May we remind you, however, that a valid license ID is required for support and that you need to be the owner of the license? Support is given promptly by e-mail or fax. When you contact us, kindly have the following information ready:

- Your license ID number and Vox Studio version number
 - A complete and detailed description of the problem
 - A written description of the command sequence that consistently produces your problem
 - Windows version number and release number
 - Complete PC description: CPU, clock speed, memory size, hard disk size, free space on all disk partitions, location of temporary directory, free space in temporary directory
 - List of other programs active while you run Vox Studio
 - Sound card type, if used, and sound driver selected in the Vox Studio tools menu.
- If applicable, have a sample sound file or script file ready.

8.3. Functional demo

A free functional demo version of Vox Studio is available directly from Xentec, from our distributors, and from various download sites. It is readily downloadable from the Internet at <http://www.xentec.be> . This demo version is fully functional but limits the length of manipulated sound files to 5 seconds and limits the number of files handled by one single batch command to 5. Other than that, the demo version is a perfect vehicle to test the capabilities and the conversion quality of Vox Studio. It is also a convincing selling tool for resellers.

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9. Glossary

Glossary of Computer Telephony Terms

[Glossary](#)

9.1. Glossary

.sd2

An extension given to ProTools SoundDesigner II files. This file extension covers several sample rates. Sd2 files are stand-alone files that have no header.

.snd

An extension given to many types of PC sound files. This generic file extension covers a variety of encoding formats and sampling rates. Snd files usually do not have a header that summarizes the information which Vox Studio needs.

.vap

An extension given to indexed voice processing telephony files. Vap files contain several concatenated ADPCM recordings and accompanying annotation text.

.vce

An extension given to NMS voice processing telephony files. This file extension covers a variety of encoding formats. Vce files are stand-alone files. They can be grouped into indexed files which usually carry the .vox extension.

.vox

An extension given to many types of voice processing telephony files. This file extension covers a variety of encoding and sampling rate formats. For example, Dialogic ".vox" files are stand-alone files and lack a header identifying the coding format and sample rate, but NMS ".vox" files actually are indexed files that contain many voice prompts and do have a header. We are sorry if this is confusing, but we did not create this mess.

.vsn

An extension given to many types of voice processing telephony files from several suppliers. This generic file extension covers a variety of encoding formats and sampling rates. Vsn files do have a header which summarizes most (but not all) of the information which Vox Studio needs.

.wav

An 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. Wav files contain a header identifying the coding format, resolution and sample rate. Wav files are to sound what Tiff files are to images.

AC

Alternating current.

ADPCM

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

A-law

The PCM companding standard used in Europe.

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 artifacts 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 wave. 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. Audio signals transmitted over the telephone network only contain frequencies from 300 to about 3400 Hz.

Audiotex

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

Batch

Method whereby many files are processed automatically, one by one, without operator intervention

Belgium

A tiny humid kingdom of 10 million inhabitants. Belgium is bordered by France, the North Sea, The Netherlands, Germany and Luxembourg. Renowned for its dark chocolate, delicate lace, hot waffles, specialty beers, good food and Vox Studio. Xentec is located in an outer suburb of Brussels, the capital of Belgium and nerve center of the European Union.

Bicom

US provider of components and tools to assemble computer telephony systems.

CCITT

Comité Consultatif International de Télégraphie et Téléphonie. Based in Geneva, Switzerland, this permanent consultative committee of the International Telecommunications Union (ITU) elaborates and publishes a variety of international telecommunications standards.

Centigram

US provider of computer telephony systems.

Compand

COMpress/exPAND, a technique to reduce the dynamic range of a signal and then restore it to (close to) its original form.

CPU

Central processing unit. The brainy part of your PC.

Creative Labs

A provider of multimedia and telephony components. Also known as Creative Technology.

Creative Technology

A provider of multimedia and telephony components. Also known as Creative Labs.

CTI

Computer-Telephony Integration. A fashionable term for what, not so long ago, used to be called simply "voice processing". CTI has become a generic term for computer-controlled or computer-assisted telephony applications.

CVSD

Continuously Variable Slope Delta modulation. A time-tested coding technique that allows representation of analog signals with less digital information than pure analog to digital conversion.

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 about the smallest change in sound volume that the human ear can distinguish.

Dialogic

US provider of components and tools to assemble computer telephony 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 dialing signals. The digits represented are 0-9, *, # and A-D. A-D are not available on standard telephones.

Elan Informatique

French provider of components and tools to assemble computer telephony systems.

filter

Manipulation that alters the spectral (frequency) components of a signal.

filtering

Process that alters the spectral (frequency) components of a signal.

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 as 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.

Group 2000

European provider of computer telephony systems.

header

A file header is a chunk of data at the beginning of a file where Vox Studio can find useful information such as recording sample rate, coding algorithm, recording length, and so on. Not all voice file formats have file headers. If Vox Studio has to open a file that has no header, it will ask the program user for the missing information it needs.

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.

InterVoice

US provider of components and tools to assemble computer telephony systems.

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.

Microlog

US provider of computer telephony systems.

Milliseconds

Thousandths of a second.

Modulation

Variation of a wave to convey a signal or information.

Mu-law

The PCM companding standard used in the US and Japan.

Natural MicroSystems

US provider of components and tools to assemble computer telephony systems.

NewVoice

US provider of components and tools to assemble computer telephony systems.

Nibble

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

NMS

Natural MicroSystems, US provider of components and tools to assemble voice processing systems.

Normalize

To make the volume of a recording as uniformly loud as possible while minimizing distortion of the sound. This is done in Vox Studio by normalizing average energy, not amplitude, of the signal.

Nortel

Canadian Telecom giant. Nortel also supplies Computer Telephony systems.

OKI

Provider of, amongst other things, silicon chips that convert an analogue signal into a variant of ADPCM. OKI chips were used on early Dialogic cards.

Pacific Image

Wrote the SuperVoice voice-mail application that comes with the PhoneBlaster card.

PCM

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

Pentium

A CPU chip used in PCs. Pentium is a trademark of Intel Corporation

Philips

Dutch Electronics giant. The Business Communication Systems Division also supplies Computer Telephony systems.

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.

Rockwell

US provider, amongst other things, of silicon chips for modem and voice applications.

Sample rate

The frequency at which samples of sound are taken.

Sampling rate

The frequency at which samples of sound are taken.

SCII

French provider of components and tools to assemble computer telephony systems.

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

SoundDesigner II

Popular Mac program from DigiDesign to produce and manipulate quality sound files.

Talking Technology

US provider of components and tools to assemble computer telephony systems.

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 or energy below which a signal causes no action or detection to take place

TouchTone

An alternative name for DTMF. TouchTone is trademark of AT&T in the US and is used by other companies in other territories.

Trailing

At the end of the sound file.

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.

Voicetek

US provider of computer telephony systems, now part of Aspect Telecommunications.

Volume

Loudness of sound signal.

VU-meter

The good-old VU-meter, usually a needle instrument, measures speech power in decibels. 1 milliwatt is the reference. The Vox Studio monitor is not calibrated in decibels and serves essentially as a visual indicator for correct recording level.

Wave

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

Windows

Windows is a trademark of Microsoft Corporation.

10. Index

About Vox Studio in Help menu	99	Demo version	123
Adjusting leaders and trailers	14	Desktop	25
Advice	114	Detecting DTMF tones	67
Amplitude normalization functionality	14	Detecting special telephony tones	66
Apple files.....	112	Devices selection.....	94
Aspect Generations formats.....	113	Dial string detection	16
Automatic recording.....	11	Dial string generation.....	15
Background conversion	55	Dialogic formats.....	106
Baseline correction	14	Dialogic indexed files	59
Basic introduction	7	Dialogic Vap file generation	59
Batch Conversion command	55	Dictionary of terms.....	128
Batch Conversion defaults	85	Digitizing a studio tape or audio CD	45
Bicom format	105	Digitizing a tape	11
Blank trimming.....	14	Disk space requirements.....	22
Bulk conversion	55	Display requirements	22
Buttons on the toolbars.....	99	Documentation formats.....	5
Buttons to record and play	100	Drag and drop loading of files	29
Calibrate in dB dBm dBV	92	Drag and drop play of files	30
Capabilities of the program	10	Dropping files in Vox Studio	29
Cascade the graphs.....	95	DTMF defaults	93
CCITT - ITU formats	105	DTMF detection functionality.....	16
Center functionality.....	14	DTMF filtering functionality.....	16
Centigram formats	106	DTMF generation functionality	15
Clean recordings to start.....	115	DTMF tone detection	67
Close a file.....	35	DTMF tone generation.....	66
Close all files	35	Editing commands	38
Configuring the prompter	44	Editing functionality.....	13
Configuring the tape loader.....	46	Elan Informatique format.....	108
Contents in the Help menu.....	97	End User License Agreement	124
Contents of user guide.....	5	Entering the license	18
Contextual menu	101	EULA text	124
Conversion and quality	118	Exit command.....	38
Conversion functionality.....	12	Expand Dialogic indexed file	61
Conversion speed and CPU.....	21	Expand NMS indexed file.....	64
Convert a file in memory	48	Fast recording of prompts	11
Convert Active command.....	52	Fast review of sound files.....	30
Convert Transform Active defaults	83	Fast visual recording of many prompts.....	43
Converting files.....	48	File format conversion.....	52
Converting many files at once	55	File formats.....	103
Copy sound to the clipboard	39	File menu	32
Copyright.....	123	File properties.....	36
CPU and speed requirements	21	File slicer	47
Create folders for Vox Studio	18	File Slicer defaults	89
Creating a new file.....	33	Folder defaults.....	81
Customize the program workspace.....	75	Format conversion.....	12, 52
Cut sound to the clipboard	39	Format for script file	103
Cutting a large multi-prompt wav file	47	Formats from the multimedia world	113
Cutting copying and pasting sound	13	Free downloaded version.....	123
dB calibration.....	92	Frequent errors.....	119
DC bias removal.....	14	Full version.....	97
Decibel calibration	92	Functional demo version.....	123
Default settings.....	18	Functionality	10
Defaults menu	78	General system requirements	20
Delete a segment of sound	39	Generate DTMF tones	66
Delete selection	39	Generating special telephony tones	65

Glossary of terms.....	128	Newvoice formats.....	110
Go to next view.....	42	Next and Sun formats.....	110
Good quality recordings.....	115	Next view.....	42
Graphical display.....	12	NMS formats.....	109
Group 2000 formats.....	108	NMS indexed files.....	62
Group and ungroup Dialogic defaults.....	90	NMS Vox file indexing.....	62
Group and ungroup functionality.....	13	NMS Vox file to standalone conversion.....	64
Group and ungroup NMS defaults.....	91	Nortel Generations formats.....	110
Group indexed Dialogic command.....	59	Objectives of Vox Studio.....	8
Group indexed NMS command.....	62	Offset removal.....	14
Hard disk requirements.....	22	OKI file formats.....	111
Headphones requirement.....	23	Open command.....	33
Help menu.....	97	Operating system requirements.....	21
High and low-pass filtering functionality.....	15	Options.....	75
How conversion affects quality.....	118	Output format defaults.....	80
IBM Directtalk formats.....	108	Overview.....	7
IMA ADPCM format.....	108	Panning in a graph.....	32
Improving speech perception.....	15	Paste sound in a file.....	39
Indexed files.....	13	Pause playback.....	72
Input format defaults.....	79	Philips Voicemanager formats.....	111
Installing the program.....	17	Philosophy behind Vox Studio.....	8
Installing Vox Studio.....	17	PhoneBlaster formats.....	111
Intelligibility Filter.....	15	Play a List of Files defaults.....	82
InterVoice formats.....	109	Play a long list of files.....	73
Introduction to Vox Studio.....	7	Play everything except selection.....	71
ITU - CCITT formats.....	105	Play the current screen selection.....	70
Judging quality.....	23	Play the current screen window.....	71
Keyboard shortcuts.....	101	Play the whole file.....	71
Legal terms.....	123	Playback functionality.....	12
License entry.....	18	Playback sound device.....	94
License ID and license key.....	97	Player buttons.....	100
License issues.....	121	Previous view.....	42
License registration.....	97	Print and return registration info.....	19
Licensing terms.....	124	Print license information.....	98
Listen to recordings.....	12	Problem solving.....	122
Loading files in Vox Studio.....	33	Program setup.....	17
Loudness of sound.....	14	Prompter command.....	43
Low and high-pass filtering functionality.....	15	Prompter defaults.....	87
Macintosh files.....	112	Prompter options.....	44
Main functionalities.....	10	Prompters menu.....	42
Main window.....	25	Proof of registration.....	98
Manual and Help file.....	5	Properties of sound file.....	36
Memory requirements.....	22	Quality issues related to conversion.....	118
Microlog Intela formats.....	109	Quality of sound card.....	114
Microphone requirements.....	23	Quality recordings.....	115
Microphone signal level.....	67	Quit Vox Studio.....	38
Microsoft ADPCM format.....	109	RAM requirements.....	22
Mixer editor and volume applets.....	115	Raw PCM formats.....	111
Monitor the input signal.....	67	Reading sound files.....	33
Move within a zoomed file.....	41	Record command.....	72
Moving in a graph.....	32	Recorder buttons.....	100
Multimedia sound support requirement.....	23	Recording a new sound file.....	72
Multi-message files.....	13	Recording files.....	11
Multi-prompt files.....	13	Recording functionality.....	11
My files do not sound right.....	119	Recording room requirements.....	24
Natural Microsystems formats.....	109	Recording sound device.....	94
New file.....	33	Recording tips.....	115

Registered trademarks.....	123	Tape loader command.....	45
Registering the license.....	97	Tape loader defaults.....	88
Registering with license key.....	18	Tape loader functionality.....	11
Registration.....	121	Tape loader options.....	46
Registration card.....	98	Technical support.....	98, 122
Removing DTMF-like sounds.....	16	Teleprompter.....	43
Removing the program from disk.....	20	Teleprompter functionality.....	11
Removing unwanted frequencies.....	15	Terminate the program.....	38
Return to previous view.....	42	Terminology.....	128
RIFF files.....	113	Test before deleting.....	71
Right click contextual menu.....	101	Test files.....	120
Rockwell formats.....	112	Third party trademarks.....	127
Sample files.....	120	Tile the graphs horizontally.....	96
Sample rate conversion.....	52	Tile the graphs vertically.....	96
Sampling frequencies old cards.....	119	Tips and techniques.....	114
Save a file.....	35	Toolbar buttons.....	99
Save all files.....	36	Tools menu.....	65
Save under a different name or format.....	35	Touch-tone detection.....	16, 67
SCII format.....	112	Touch-tone filtering.....	16
Screen resolution.....	22	Touch-tone generation.....	15, 66
Script file format.....	103	Trademark acknowledgements.....	127
Scroll a file.....	41	Transform Active file.....	48
Scroll left.....	42	Undo previous commands.....	38
Scroll right.....	42	Ungroup indexed Dialogic command.....	61
Scrolling in a graph.....	32	Ungroup indexed NMS command.....	64
Search in the Help file.....	97	Uninstalling.....	20
Select the whole sound file.....	40	Unregistered version.....	123
Selecting a segment of sound.....	30	Upgrading a license.....	121
Set defaults.....	18	User interface.....	25
Silence adjustment functionality.....	14	Using toolbar buttons and mouse clicks.....	99
Silence while recording.....	24	Vap file to standalone conversion.....	61
Slicing a wav file in chunks.....	47	View less details.....	41
Slicing an existing wav file.....	11	View menu.....	40
Solving problems.....	98	View more details.....	41
Sound card quality.....	114	View the whole file.....	41
Sound card requirements.....	23	Viewing the input signal.....	67
Sound card selection.....	94	Visualizing the sound files.....	12
Sound file formats.....	104	Voice talent requirements.....	24
Sound input device.....	23	Voicetek Generations formats.....	113
Sound is not as expected.....	119	Volume control for record & playback.....	115
Sound level adjustment.....	115	Volume normalization.....	14
Sound level normalization.....	14	Wave file formats.....	113
Sound Mapper.....	94	Waveform display functionality.....	12
SoundDesigner II format.....	112	Window maximized and minimized.....	28
Speaker requirements.....	24	Window menu.....	95
Stop playback.....	72	Windows limits for zooming.....	40
Studio requirements.....	24	Windows requirements.....	21
Studio sample files.....	120	Windows wav formats.....	113
Summary of menu commands.....	25	Workspace.....	25
Sun and Next formats.....	110	Zoom all.....	41
Support.....	98, 122	Zoom in.....	41
Support section.....	121	Zoom out.....	41
Table of contents.....	5	Zooming in and out.....	31
Talk-off reduction.....	16		