Speakerphone 3 manual v1.0

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Introduction

A bad GSM connection on a busy sidewalk, bullhorn with feedback and helicopters overhead, or a 1952 rockabilly guitar amp in a recording studio's live room, Speakerphone gives you authentic speakers of any size together with their natural environments.
Speakerphone, powered by Altiverb, combines a wealth of effects including a convolution engine that uses actual samples of hundreds of original speakers, a radio receiver tuning dial, record player scratch and static generator, GSM cell phone data compression, distortion, tremolo, delay, a variety of EQ and dynamics, bit crushing, sample rate reduction, a full blown convolution reverb, and a library of samples to combine into entire environments.
An impulse-response based speaker emulator, a convolution reverb and a sample player all in one: Speakerphone can produce complete scenery where any speaker is involved.

An obvious application for Speakerphone is televisionand film audio Post production. Speakerphone will give you all the walkie talkies, distant transistor radios, upstairs TV sets, bullhorns and cell phones you'll ever need. It will add dial tones, operators and static, and you can select from a wealth of environments on either the caller's or receiver's end. It'll make your voice sound through a distorted megaphone suspended from the ceiling of a station hall, while mixing in passenger's footsteps around you. And you can simply drag anything from the sample bay right to your Pro Tools track and vise versa.

The end-all speaker simulator plug-in, with a host of environments to put them in.

Basic Operation

Insert Speakerphone on a track of your audio workstation software , and playback dialogue, or speak live through the track. Make sure the input level is such that the inpulevel indicator is in the green area. That's where balance between extra sounds and input is right and all the distortions and gates and such operate as intended. You can select a preset from one of the many categories in the browser. just click on the preset name at the top left to bring it up.

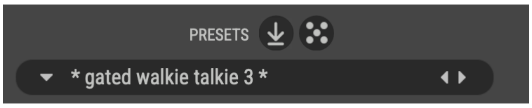
If you found something close, but you need it a bit different start tweaking the controls. A module is active when its power control lights green. Each module is described in the next chapter. If you see no controls you can show them from the 'expand' menu at the top left. That's where you resize Speakerphone as well.

CTRL or right -click on a control, such as a knob, to bring up a menu that allows you to attach a parameter to a MIDI controller or an LFO. The knob will blink until a controller is sent on channel 1
The heart of the preset is the Speaker module. Click on the phone icon to select one of many phones, or on the radio icon to select a radio .

The background ambiences that you hear in many presets come from the Sample Bay at the bottom. Click on a sample to start it, on the fader handle to adjust its volume, on the green-lit triangle to stop, and on the rotary control to adjust panning. Click on the name of a sample pack ("kids" in this example) to select another sample pack.
Suppose the voice (or your complete mix) now sounds from a radio. If you play back stereo music through this setting then perhaps at some point you want to slowly 'open up' the sound to your full bandwidth stereo input sound. Click he DRY button to make the slider move from wet to dry, from treated audio to untreated audio. Or drag the WET/DRY slider manually.

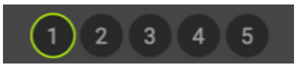
You can drag files between the Finder/Windows, the sample bay and your tracks, so if you like the ambience sounds, or any other sample from the sample bay, but you'd rather have it on your track than coming from Speakerphone, you can click it, hold it for a second and then drag the sound from the sample bay to the track.
Use the large Input and Output controls to adjust the volumes of incoming and outgoing audio of Speakerphone. beneath the OUT control there is a limiter. 'Mute on stop' means playback of samples from the sample bay will be paused when your DAW is not playing.

Presets



Click on the preset name. A browser shows up. You can select another preset now.  saves a preset in the 'User' catagory.  selects a random preset.

Automation presets



A preset can also be stored in one of 10 numbered circles in the top bar. Store such an 'automation preset' by clicking on an empty slot, one that is still dark as in the picture above. Recall a preset by clicking on the filled slot. 'automation preset index' is an automatable parameter. You can therefore invoke switches using your workstation's parameter automation. The contents of all automation preset slots are in turn saved in a single (Pro Tools) Preset, as well as in the Session file. Please not that the entire sample-bay's state is in a preset as well as an automation preset. so if you need the samplebay, the automation preset is the best way to automate. If you do that we advise to take all other parameters out of automation so you are sure the automation preset is always boss. on Mac **CMD-click on an occupied slot erases its contents.** On Windows Alt-click.

Wet / Dry control

The Wet Dry control gradually moves between processed and unprocessed (Bypassed) sound in an elaborate way. A movement from WET to DRY will gradually open up filters, including the speaker IR, and diminish the effects of modules like Distortion and Telecom each in a unique way. The overall result of a movement from WET to DRY is a seamless 'opening' up of the sound from the complete preset to unprocessed sound.

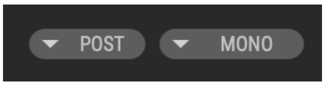
The WET DRY SLIDER changes all parameters from their current setting to 'through'
The dry button starts to move the WET/DRY Slider gradually from wet to dry
The turning knob next to the slider determines how fast the WET/DRY slider will move once the DRY or WET buttons are clicked
This WET button starts to move the WET/DRY Slider gradually from dry to wet
If, for instance, you have your mix sounding from a jukebox in the back of a bar, you can slowly 'open up' the sound to full stereo by dragging the WET/DRY slider. Or click the DRY button to make the slider move from wet to dry, automatically, at a speed determined by the rotary control to the right of the slider.

Speaker

Selection of the speaker impulse response.

At the heart of the speaker simulator lies a library of recordings of different speakers, often referred to as Impulse Responses, or IR's. These impulse responses by themselves capture many characteristics, both in frequency and time, of the sampled speaker. The process is called convolution. The speaker module lets you select a speaker impulse response, and it shows its corresponding photograph. Double click the photograph to get additional info on the selected Impulse Response.

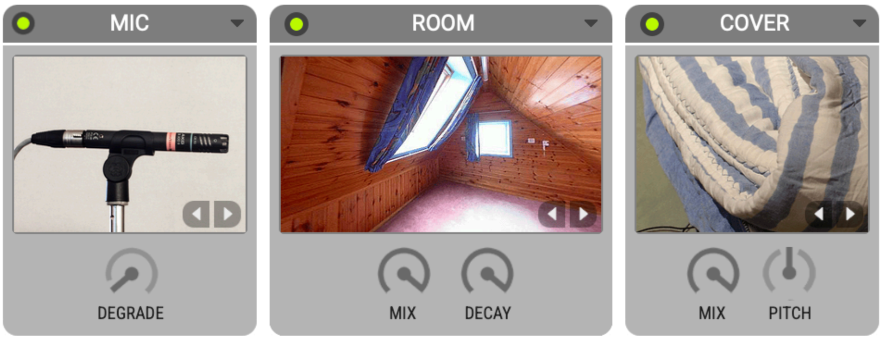
When you click a speaker category icon, like the phone or the radio, a browser window appears that lets you select a speaker in that category. To make auditioning as quick as possible, the sounding selection is continually updated while you navigate the browser. Just hold down the mouse button while hovering over the items in the browser. The picture in the circle shows the actual speaker that was sampled.



Pre/Post -Selects one of two positions in the signal chain where the speaker module can be inserted. The graph below shows both positions. Suppose a selected speaker has a frequency range up to 5000 Hz. Using pre-convolution Distortion and crunch will add harmonics above 5000 Hz in the end result. If you want those to be cut-off by the IR, use post-convolution, shown in the connection of modules below.

```
inputgain > sample bay (if pre) > phono > tuning > telecom > speaker (if pre) > crush >
gate > compressor > distortion > mod > delay > mic > speaker (if post) > Leslie > cover
> room > eq > sample bay (if post) > out > limiter
```

Stereo/mono in stereo mode the IR will be used twice to treat the input: one for the left input channel, and the other for the right. In mono mode, input channels will be summed before they enter the speaker module. This way you can set up a mono tube-radio patch and run your stereo mix through it. When you use the dry/ wet slider above, the signal will go from mono to stereo as well as from tube radio to neutral.



Mic Selection of the microphone impulse response.

Similar to the speaker selector, but here a microphone impulse response can be selected. The available microphone impulse responses capture time and frequency characteristics of the microphones depicted, and therefore make it sound as if the selected microphone was used in the result. Use it to make an announcer speak through a typical announcer microphone, or to pick up a guitar cabinet using a classic microphone for the purpose, for instance a Royer.

DEGRADE - in old (mainly telephone) carbon microphones a particular type of degradation is simulated with this knob. Turn to right for the crackly effect

Room

Convolution reverb for room, outdoor space, hall, resonant enclosure, and reverb gear simulation.

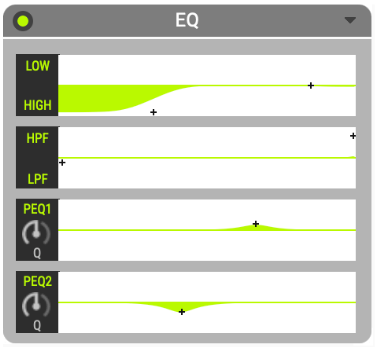
Room creates reverb based on recordings of actual spaces ranging from a railway station hall to the cockpit of a MIG fighter plane. Samples of springand plate reverbs are available as well. Click on the photograph to select a different acoustic space to place the speaker in.

MIX turn to the right to get more reverb, or to the left to add more dry audio.
DECAY turned all the way clockwise the decay, or reverb tail length, is complete. Turn it counter clockwise to shorten the reverb tail.

Cover

cover up the source sound with blankets, boxes, trunks

A variety of objects can cover a sound source. Blankets, suitcases, boxes, glass cups, car trunks are just few. Click on the picture to select such a cover. All of these are actual impulse response recordings in Speakerphone.
MIX turn to the right to get more of the cover's characteristics, or to the left to add more dry audio.
PITCH often the covers have very distinct resonances. Tune them using this knob. Try it on a wine or beer bottle.



Distortion

Emulate a speaker's or amplifier's over modulation.

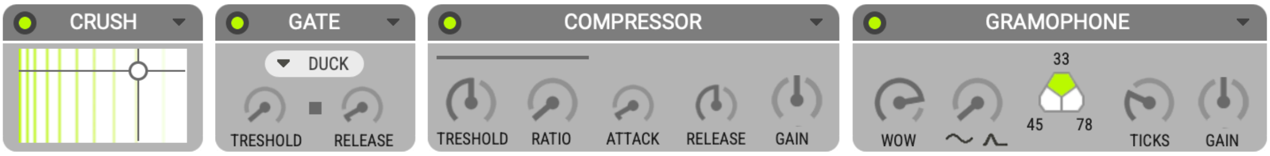
In the distortion module audio is first passed through the Pre EQ. If you engage it by clicking its on/off button you can control both a parametric EQ and a low pass filter with resonance in the same graph. Bandwidth, or Q, for the Parametric EQ can be adjusted with the small rotary control.
Next, the signal hits the PRE gain, which typically amplifies so the waveform will become more or less distorted in the next module: the Distortion Type.
The actual distortion model is selected in the 'TYPE' popup. Some are amp models, others are wave-shapers, all of them are further controlled using the CURVE graph at the bottom of the distortion module.
POST-gain controls the distortion's output volume. You can also click the connection-button in between PRE and POST to have post gain automatically decrease when pre gain increases and vise versa.
Finally you can Mix the distorted signal with the undistorted input signal using the MIX control.
The effects of Distortion are different when it sits before or after the Speaker module. The position of the Speaker module can be adjusted by the Speaker's PRE/POST control (see Speaker section).

Equalizer section

5 types of frequency filters.

A filter can be turned on or off by clicking its name. Low and High shelving filters Click on a cross in the graph and move it up to amplify, down to attenuate and left-right to change the cut-off frequency.

High pass filter and low pass filter with separate resonators Click and drag in the graph to move the cross hairs. Left and right
adjusts cut-off frequencies, up and down adjusts resonance amount. Resonance is a means of emphasizing the cutoff frequencies themselves. The high pass and low pass filter's frequency responses do not flatten out like the shelving filters, making them the more dramatic choice of the two.
PEQ1 AND PEQ2 - Two bands of parametric Equalization
Horizontal movement adjusts the position of the notch or peak. The amount of amplification or attenuation is controlled by vertical movement. The width (q-factor) of the peak or the notch is adjusted with the rotary knob.



Crush word length reduction and sampling rate reduction.

Drag the cross hair down to reduce word length (reduce from 32 bit input to 2 bit output). Move to the left to reduce sampling rate.

Gate Mutes or attenuates the signal when it’s below the threshold level

TRESH Threshold the gate mutes or attenuates the
REL - ‘release’ - how quickly the gate returns to no gain reduction once the input level gets above the threshold.
MUTE vs **ATTENUATE** - Select in this popup menu wether the gate completely mutes or just attenuates the signal.

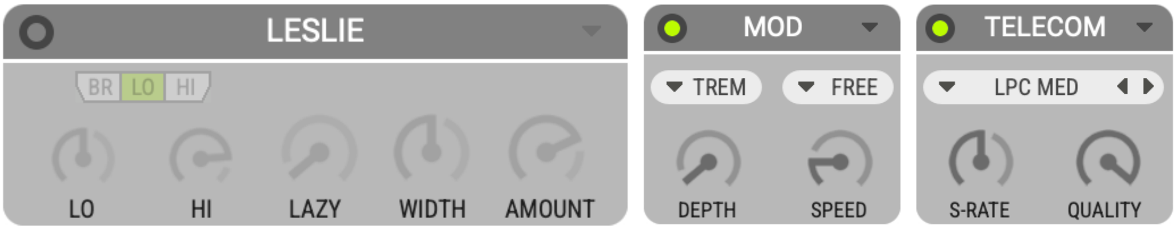
Compressor Dynamic compression controls.

The horizontal meter continually displays the amount of gain reduction. Note that limiter near the ‘OUT” control at the right top right.
TRESH Threshold - the compressor reduces the dynamic range of the incoming audio signal if it becomes louder than this
RATIO The amount of gain reduction. 4:1 means 4 dB of additional input creates just 1 dB of additional output level

ATK attack - how fast the compressor responds to changes in input level.
REL release - how quickly the compressor returns to no-gain-reduction once the input level falls below the threshold.
GAIN - Volume of the output.

Gramophone Gramophone effects simulator.

WOW AMT - The hole in a record can be off center. The further the groove is from the hole, the faster it will be moved beneath the stylus, causing the pitch to swing up and down. AMT controls how far the hole is off center.
CURVE - A pitch curve also occurs when the record is not flat. You can change the shape of the pitch curve from a perfect sine (result of an off-center hole) to a steep peak every turn (result of a steep bump in the record).
33/45/78 RPM - The number of record rotations per minute. Influences the speed of the curve as well as the speed of repetitions in the TICKS parameters
TICKS The density of the layer of crackles and ticks.
GAIN The gain of the layer of crackles and ticks.



Leslie Emulation of the classic rotating speaker

This module emulates a Leslie speaker, typically associated with Hammond organs.
A Leslie speaker, on the inside, rotates a horn speaker for the mid to high frequencies, and it rotates a horizontal baffle below a downward facing low frequency woofer. The result is a rich rotating sound.
For authentic results first select from the speaker module a Leslie speaker. You'll find it in the miscellaneous speakers section. Then use the following Leslie controls to rotate it:

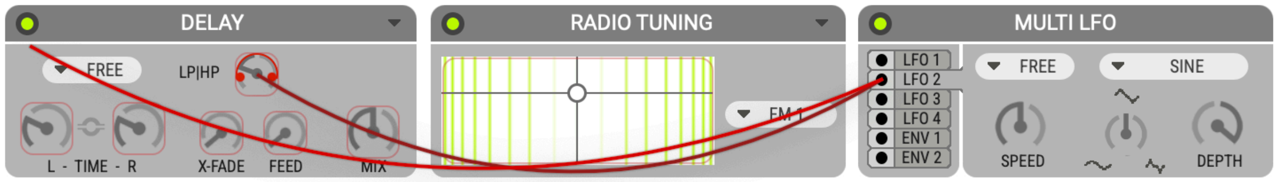
HI AND LO KEYS switch the rotating elements between two speeds. The corresponding speeds can be set using the LO switch (brake). AND HI KNOBS. The rotation can be stopped using the BR
LAZY determines the the time the rotation speed needs to reach HI or LO.
WIDTH Controls the stereo width of the rotation effect.
AMT Amount controls the intensity of the Leslie's effect. Together with 'Width' it would in reality be achieved by microphone positioning.

Mod Tremolo, Chorus, Phaser, Flanger or Vibrato.

Mod offers 5 types of classic modulation effects, available in the top left popup.
DEPTH Controls the amount of the modulation effect.
SPEED Controls how fast the modulation oscillates. Speed can either be set in FREE mode or in SYNC mode. SYNC ties speed to the host sequencer's tempo and displays it with a note value. While FREE offers a rotary knob.

Telecom (Cellular) phone connection protocol (degradation) simulation.

This module converts the audio to actual telecommunication industry standard audio data compression such as 'GSM', and decodes it back to audio, in between dropping data on purpose.
This module faithfully reproduces the typical audio artifacts found in bad phone connections. Repetitions, underwater effects, it will all sound very familiar. As a side effect of the LPC coder/decoder (which is in fact a voice re-synthesizer in actual phone connections) the Telecom module doubles as a classic robot-harmonizer. You can build synthetic voice effects using LPC fixed pitch, or even send it MIDI notes on channel 3 to play pitches.
choose the encoding/decoding type of data compression that is used in actual (cellular) phone connections. Some of them may introduce latency (delay) in the same way actual telephone lines do. The 'Liquid' degradation types introduce the smallest delay.
SRATE changes the sampling rate of the phone data transmission.
QLTY quality simulates the effects of a bad connection. Most notably dropped (and therefore repeated) time-slices. In Fixed Pitch mode the QLTY knob controls pitch of the robot voice.



Delay mono or stereo feedback delay with filter.

STEREO MONO - switches the amount of output delay channels.
TIME L left channel delay time.
TIME R right channel delay time.
FDBCK amount of the delay output that is fed back into the delay input. LP/HP an optional low pass and high pass filter in the feedback path. Turn to
XFD Cross feed determines how much of the left channel signal gets fed into the right channel delay path.
MIX determines the mix between the delayed signals and the input signal.
FREE mode lets you adjust delay times in milliseconds, while
SYNC mode locks the timing of the delay to the host sequencer's tempo. The delay time rotary knobs are exchanged by notes, and clicking on the note brings up a note-value selector box.

Radio Tuning Radio receiver tuning.

Simulates the various effects of a radio receiver dial.
STEREO MONO - in mono mode the tuner's output is summed to mono
VERTICAL MOVEMENT changes the frequency of inter-modulation side tones.
HORIZONTAL MOVEMENT when the cross hair is moved out of the center reception of the signal worsens (increasingly adds distortion and noise).
MONO/STEREO in stereo mode both the band-noise and the swooshes are different on the left and right output channel.
RADIO BAND SELECTOR POPUP (FM/AM) Different radio bands give different types of band-noise, modulation and distortion effects. Select the type radio band here.
Please note that there are three sample-pack's in the radio/megaphone category that are dedicated to authentic static and wooshes of radio bands.

Multi LFO

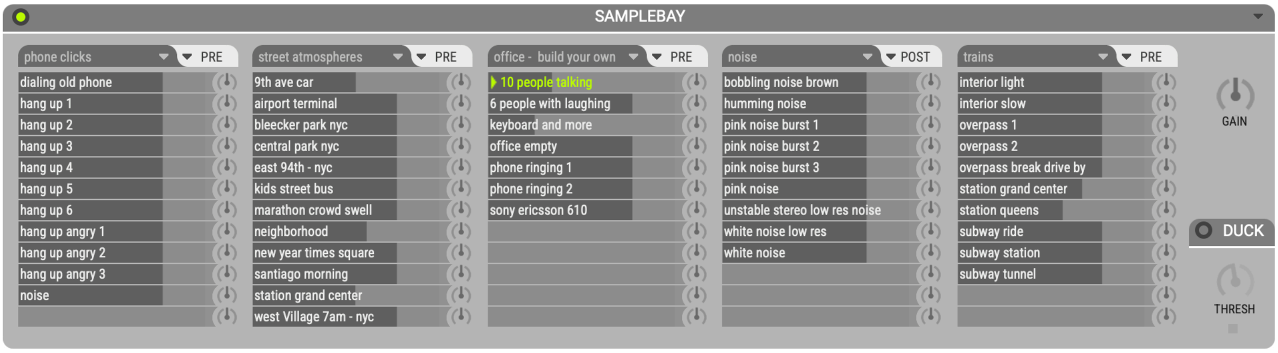
4 Low Frequency Oscillators and 2 envelope followers for automated parameter control.

Connecting and adjusting an LFO

Click and hold the black dot before a line that says 'LFO'. Then drag towards any on screen control, such as Leslie AMOUNT. Then release the mouse knob. You have now connected lfo 1 to Leslie AMOUNT. Adjust the minimum and maximum value that the LFO reaches by dragging the red limitation points around the Leslie AMOUNT knob. Further adjust the shape of the lfo by selecting a shape from the shape popup (where it says 'sine'), and use the shape turning knob to distribute the movement to the far ends of the cycle, or to the middle. Speed (lfo frequency) can be set freely, or locked to the beat, in which case note values are shown.
Instead of dragging a cable between the lfo and the parameter you can also right click on a parameter and use the pop-up menu to connect to an lfo or envelope follower.

Connecting and adjusting an envelope follower
Attack and release determine how quick the envelope follower reacts.
Connecting is done similar to an LFO. An envelope follower increases the value of the connected parameter according to the level of incoming audio. It starts reacting when audio exceeds the lo-threshold, and it will not increase further when the level exceeds hi-threshold.

Disconnecting an LFO or envelope follower
This is done by right clicking (or CTRL clicking) the parameter it controls, and selecting "off" such as in the picture to the left. This pop up menu also offers the alternative way of connecting a parameter to an LFO or envelope follower.



Sample bay A library of samples for playback by mouse clicks or MIDI

You can build complete environments for the speakers here. Several gigabytes of samples and music are installed with Speakerphone, free for use in your (commercial use) productions. 5 tables of twelve samples are accessible at a time for playback via mouse clicks or MIDI. The samples range from extras like knobs and closing car doors to ambiences and music, and they are used throughout the presets that come with Speakerphone.
A sample can be played by clicking on it, or dragged to the Finder (where it will appear as a sound file) or to the tracks of your sequencer.
MIDI consult the manual of your sequencer to find out how to send MIDI notes from your MIDI keyboard or sequencer tracks to a plug-in. The sample Bay receives MIDI notes and volume control on Channel 2. The first sample-table runs from MIDI note number 24 (c-2) to 35 (b-2) The next table is triggered by the next octave and so on.
(Alt) drag samples from sample bay to tracks. You can also drag samples to the Finder/Windows, or between slots within the samplebay.
You can move the samples folder to another drive, but after that you will have to reconnect Speakerphone by selecting 'choose samples folder' from the arrow in the top right of the sample bay.
Select wether the sample is played before (PRE) or after (POST) effects. So here you decide wether we receive a call IN a car, or we receive a call FROM a car.

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```

Click on a sample to play. Many of them are seamless loops.
Drag the horizontal fader to change gain.

Click on the heading name of a Sample Pack to select one of many sample packs.
Switch on 'Duck' if you wan the samples to be 'ducked' by your input audio, like a radio DJ ducks music when he speaks.