SIPob

Installation & Configuration Guide

Version 1.3

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https://www.kaplansoft.com/

SIPob is built by Yasin KAPLAN

Read "Readme.txt" for last minute changes and updates, which can be found under the application directory.

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Introduction

SIPob is an outbound dialer based on RFC 3261 to make telephone calls to a set of user defined numbers. SIPob executes user defined scenarios when remote party answers. You can select your own audio files to be used in a scenario *(They must be a wav file in 16 bit per sample, 1 channel and 8000 Hz sampling frequency)*. SIPob can also read-out texts using TTS *(Text-to-Speech)* engine

SIPob supports NAT traversal. You can log session details into a log file and monitor active sessions in real-time.

SIPob also supports UPnP IGD specification. If it is installed behind an UPnP supported Internet gateway device *(e.g., ADSL router)*, SIPob automatically detects if it is behind a new NAT gateway and its external IP address. All outgoing requests are manipulated for NAT traversal. You do not need to add manual reverse mappings for SIP or RTP protocols.

SIPob support UDP, TCP and TLS transports.

System Requirements

1. A Windows system with at least 2048 MBytes of RAM.
2. Microsoft.NET Framework v4.0 *(Min.)*
3. 10 MBytes of disk space for installation.
4. Administrative privileges.

Installation

Unzip "SIPob.zip" and click the "Setup.exe" that comes with the distribution. Follow the instructions of the setup wizard. Setup will install SIPob and add a shortcut for SIPob to the desktop and the start menu.

Configuration

Run SIPob Manager from Start Menu / Program Files / SIPob. SIPob automatically configures itself at first run.

SIPob also checks if it is installed behind an UPnP supported NAT gateway. If so, SIPob automatically detects the external IP and displays it on the status bar. SIPob also adds a reverse mapping for incoming UDP connections automatically *(Default UDP port 5060)*.

Settings Tab

Click Settings Tab to start configuration. The settings tab has four sub sections.

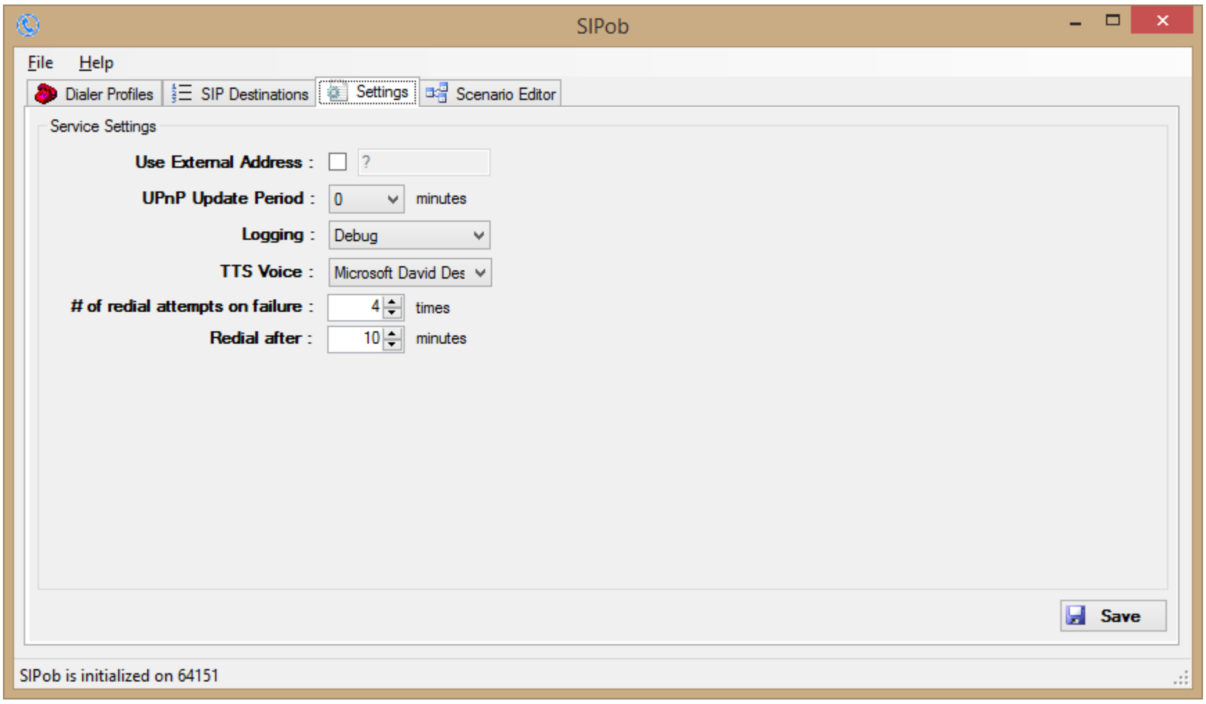


Figure - 1. SIPob Settings / Service Parameters

Enter the following information for the Service Parameters:

- **Use External Address:** If SIPob is installed behind a NAT gateway which does not support UPnP, you can set external the IP address manually for NAT traversal. If your NAT gateway supports UPnP, set the UPnP Update Period to value greater than "0". You can specify a FQDN *(DynDNS address etc.)* as an external address; SIPob will query FQDN every minute for possible IP address changes.
- **UPnP Update Period:** You can specify the period for querying the UPnP Internet Access Gateway. Set to "0" to disable UPnP support.
- **Logging:** Select the logging level of SIPob. Select "None" if you do not want logging, select "Errors" to log errors, and select "Sessions" to log session information and errors. Log files are located under the <Application Directory>\Logs directory.
- **TTS Voice:** SIPob can synthesis audio based on the text entered in Audio tab. Select TTS voice profile for audio synthesis.
- **# of redial attempts on failure:** Set how many times SIPob will try to make a call to a failed destination after the first attempt.
- **Redial after:** Specify time in seconds, how much time will SIPob wait between re-dial attempts.

If you click the [Save] button, the settings will be saved to SIPob.ini.

Routing

You can define static routes to SIP endpoints through the "SIP Destinations" tab. Enter a phone number prefix to bottom leftmost textbox, enter the endpoint IP address to the textbox to the right of the prefix entry. You can also enter the SIP port *(Default 5060)* used by the SIP Endpoint and the Endpoint type *(Default SIP UA)*.

You can also have a default route entry as shown the figure below. SIPob chooses the longest match prefix route. If any match cannot be found, the default route is chosen if it exists. If the next hop configured for a phone prefix requires authentication, you can specify a username and password for the particular routing entry. If authentication is not required, you can leave the username and password fields blank.

Enter a prefix and click the “Add Route” button to add a new routing entry. You must edit at least the Gateway entry to be able to commit the changes. You can specify a separate domain name if the domain name is different to the Gateway IP address or the FQDN. If the configured route requires TCP transport, you can set it by the **Transport** parameter. If you set Remove Prefix = Yes, SIPob will remove defined prefix from the dialed number. You can specify trunk capacity for the defined SIP routing entry. SIPob will queue calls to this destination when the capacity is full.

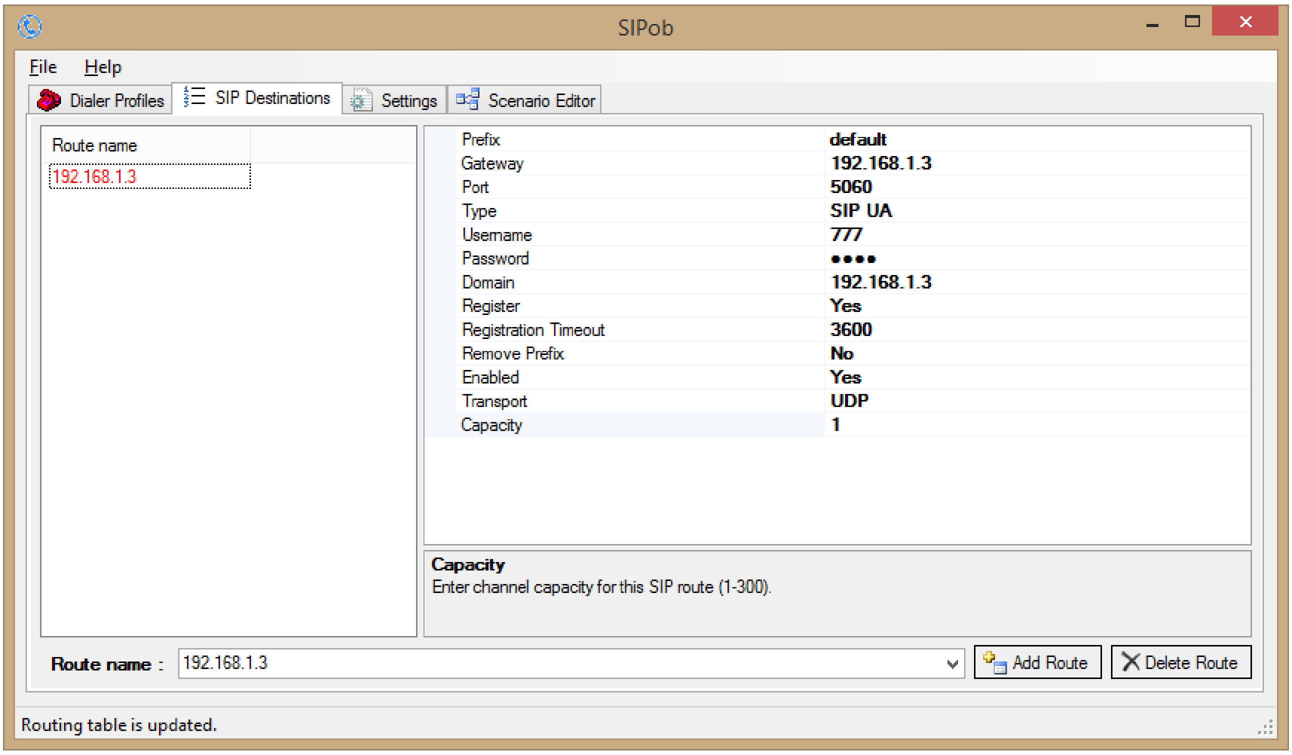


Figure - 2. SIP Destinations Tab

Scenario Editor

You can create your own scenarios using built-in Scenario Editor. Before creating your scenario, you must define audio prompts to be used in your scenario. Prompt and Action definitions for the scenarios are stored in SIPob.db.

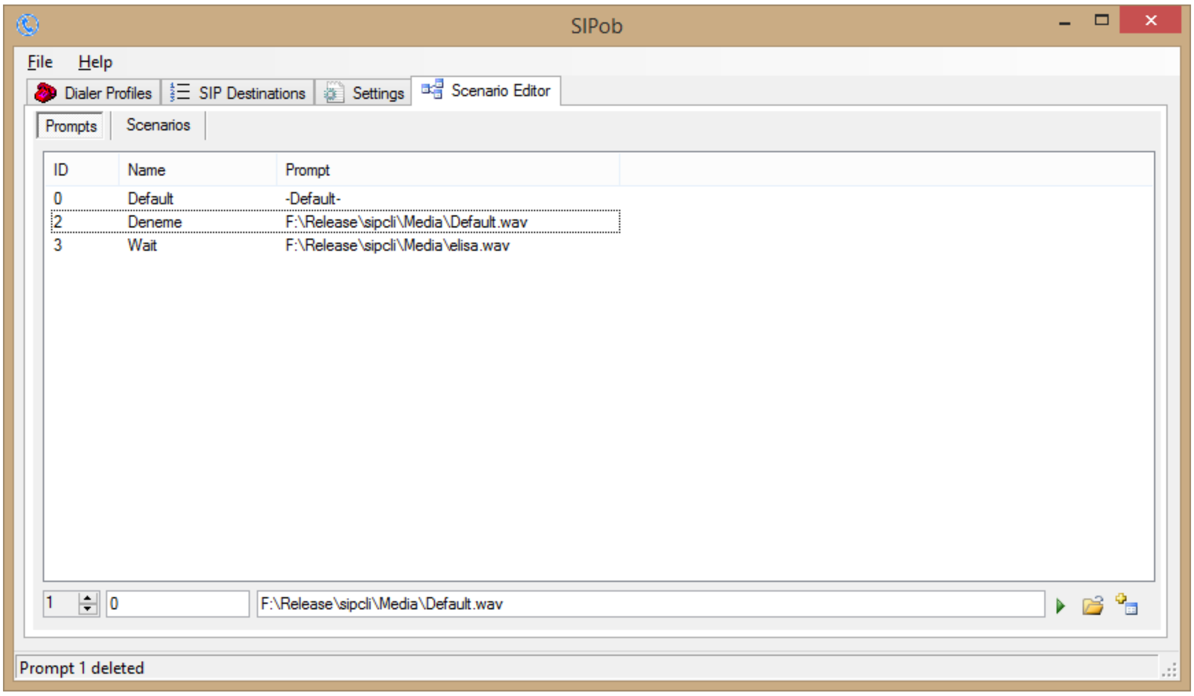


Figure - 3. Scenario Editor / Prompts Tab

Scenario Editor / Prompts Tab

Click Scenario / Prompts Tab to add audio prompts. You can either add wave files in 16 bit per sample, 1 channel and 8000 Hz sampling frequency format or free text to be played out using TTS engine. Each prompt must have a **unique** Prompt ID. You must also a descriptive name for the prompt. "-Default-" specifies built-in welcome announce. You can delete a defined prompt pressing delete key on the keyboard. Scenario Editor does not allow deleting a prompt which is used in a "Play" action.

Scenario Editor / Scenarios Tab

Click Scenario / Scenarios Tab to define scenarios and steps in your scenarios. Each action must have a **unique** Action ID in a scenario. You can delete a defined scenario or action pressing delete key on the keyboard after selecting the item to be deleted. You can specify called numbers and time frame when the scenario invoked by SIPob. You should also specify maximum number of steps can executed in scenario to prevent infinite loops. Scenario Editor does not allow deleting an action which is used in an action as a *NextAction* and you cannot assign Action's *NextAction* to itself. SIPob supports following actions in the scenarios:

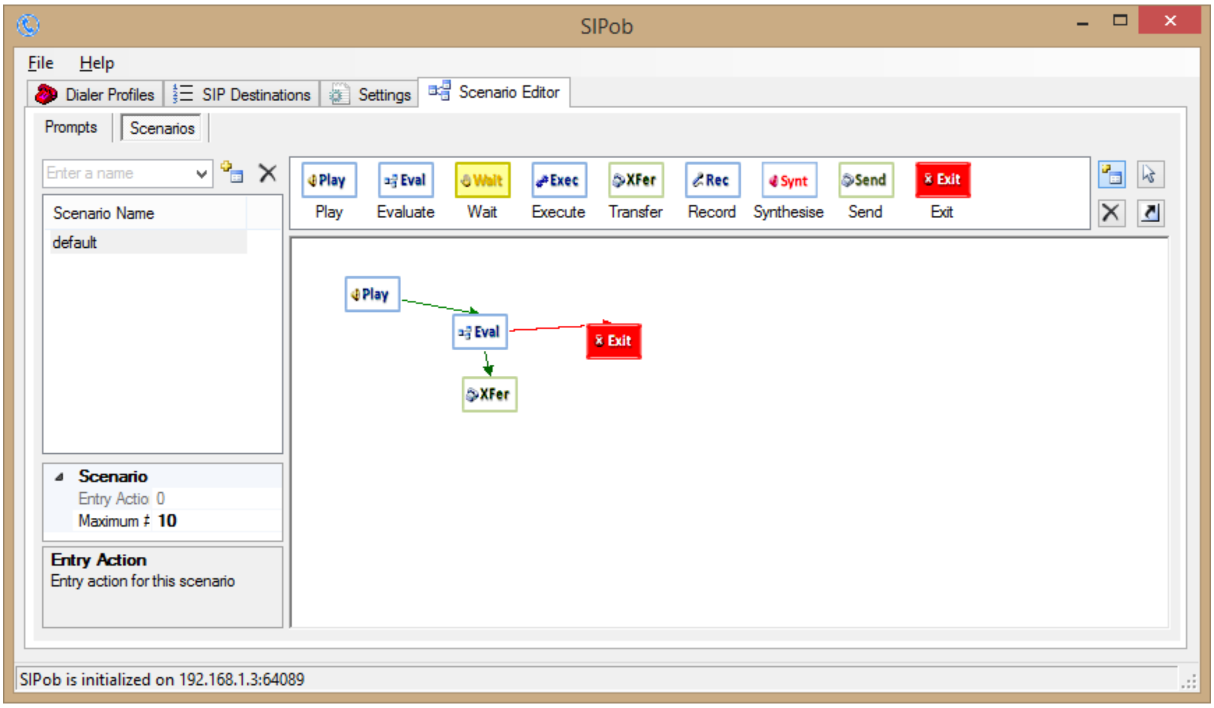


Figure - 4. Scenario Editor / Scenarios Tab

Play. Play action will play out defined in action *(Prompt)* and collects user input in %received_digits% variable. **Variables are case sensitive.** You can set how many times welcome message will be played if no user input *(DTMF digits)* detected *(Count)*, how many digits will be collected maximum *(DigitLength)* and how many seconds will be waited after last digit entered if user does not dial # *(DigitTimeout)*. You must also define next step after the action completed *(NextAction)*.

Synthesise. This action is like Play action and has the same parameters. However, Synt synthesises audio prompt dynamically using Microsoft TTS engine by processing test string specified in Prompt parameter. You can use internal variables in prompt text like %called_number%, %calling_number%, %received_digits%, %date%, %execout% and %sipdomain%. **Variables are case sensitive.** <u>You cannot use pipe | character in prompt text.</u>

Record. Record action will record audio from calling part after playing out a prompt which is configurable. You can restrict maximum duration of the audio recording by Duration parameter. Caller can terminate recording by dialing pound key *(#)*. You must also define next step after the action completed *(NextAction)*.

Wait. You can wait caller while playing out a configurable prompt. You can specify wait time in seconds. You must also define next step after the action completed *(NextAction)*.

Evaluate. Evaluate action evaluates user input and then determines next action. If SIPob cannot finds an action satisfies conditions in *Options* parameter next action will be the action defined in *DefaultAction* parameter. Conditions must be entered in <Go to Action>;<Case 1>|<Go to Action>;..;<Case n>|<Go to Action> format.

Execute. Execute action executes the executable defined in *ExecutablePath* parameter. You must enter executable with full path. Valid variables for the command line parameters *(ExecutableParameters)* are %called_number%, %calling_number%, %received_digits%, %date%, %execout% and %sipdomain%. Variables are case sensitive. Leave *ExecutableParameters* blank if no parameter will be used. Set WaitforCompletion > 0 in seconds if you wish SIPob to wait completion of the execution of the executable and SIPob will use return value (DOS Errorlevel) of the executable as the next action be executed. You must also define next step after the action completed *(NextAction)*. If you set WaitforCompletion > 0 SIPob will store executable's console output to %execout% variable and ignores return value *(DOS Errorlevel)* of executable.

If you set Executable path to `%httpget%` and enter a URL as executable parameter, SIPob will connect to the URL, get response and will set `%execout%` variable to the web server response.

Transfer. Transfer action will transfer the call to the extension defined in *Number* parameter. You can have a predefined extension or `%called_number%`, `%calling_number%`, `%received_digits%` *(User input)*, `%date%`, `%execout%` and `%sipdomain%` variables in *Number* parameter. Variables are case sensitive. You can specify a prompt to be played out while transferring the active call. You can also specify a next action for failed and successful call transfers.

Send. SIPob can send DTMF digits to the remote party

Exit. Exit action terminates execution of a scenario. You can specify a prompt to be played out while terminating the active call. If any of the action has an undefined action in NextAction parameter SIPob will also terminate execution of the scenario.

It is wise to start your scenario with a Play action.

You can add an action to your scenario by dragging it from Actions list on left after clicking “Add action” button. You can assign an Action’s next action by dragging it to another action after clicking “Link action” button. You can change the next action in the same way. You can delete an action or a link after clicking “Delete object” button and then selecting object.

You can set startup action by double clicking an action in the scenario map.

Dialer Profiles

You can create dialer lists through Dialer Profiles tab. You can have independent dialer profiles to be executed user defined date and times. You can either enter phone numbers to a profile manually or you can import a text file which contains phone number entries.

You can start a dialer profile by clicking Play button. SIPob will queue calls if you specify a Start Date / Time greater than current Date / Time. You can monitor call status real time and hang up or force to re-queue a failed call.

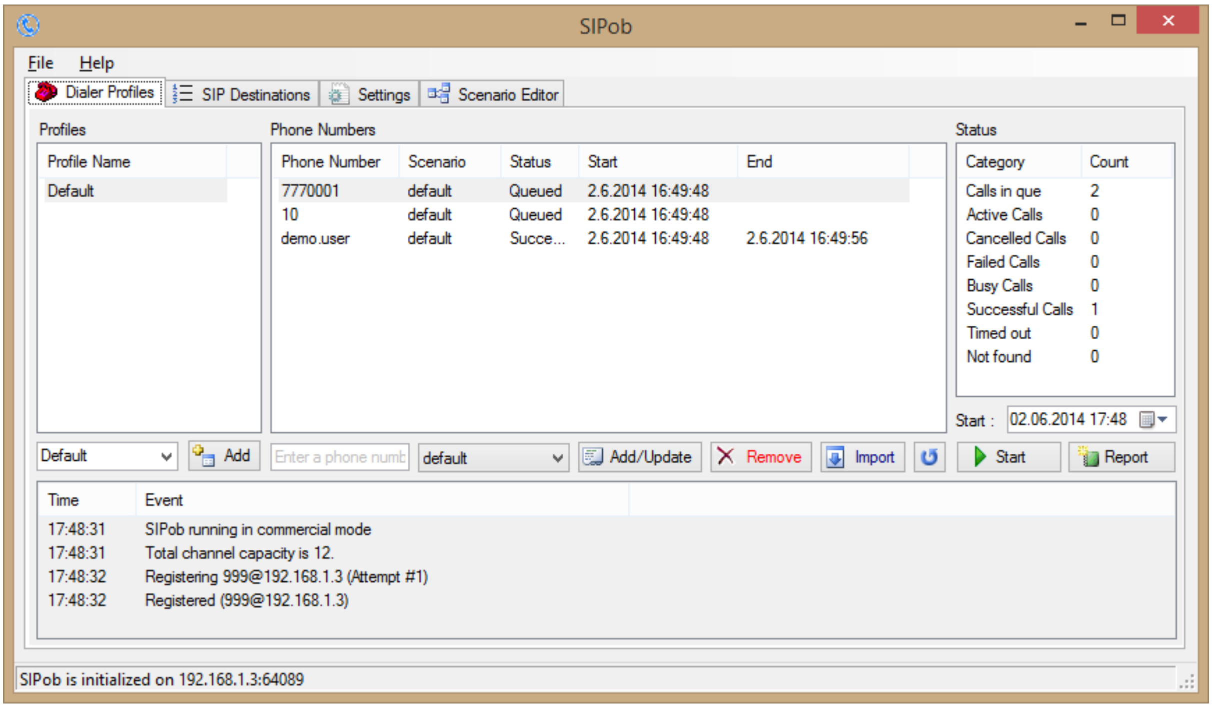


Figure - 5. Dialer Profiles Tab

If the SIPob dialer profiler cannot be started, please examine the Application Log tab as well as the SIPob log file under <Application Directory>\Logs, ensuring that you have enabled logging in “Settings/Service Parameters”. Log file is also accessible through file menu.

Troubleshooting

SIPob provides many messages when problems occur. You can see error messages on the SIPob Status bar or in the log file of the SIPob service. You can enable logging in the Settings Tab. There are three levels of logging: None, Errors, and Sessions. If you select “Errors”, SIPob logs just error messages. If you select “Sessions”, both Session and Error messages will be logged. You have to save or apply settings changes if you change the logging level setting. Log files are located under the <Application Directory>\Logs directory.

SIPob Messages

`Unable to initialize UDP/TCP thread [x.x.x.x:5060]`

If another application is configured to use the same UDP/TCP port *(5060)* as SIPob, SIPob cannot initialize the respective thread.

`Default route points to this host`

You cannot specify a gateway points to SIPob.

`New setting(s) applied and activated. Check default route.`

There is a problem with the IP address or FQDN of the default route.

`Cannot apply changes; enter minimum configuration`

There is missing configuration data.

`You cannot redirect an endpoint to itself.`

You cannot re-direct an endpoint to itself.

`Invalid endpoint information or illegal character detected in entries.`

Invalid characters found in a SIP username or entry. You can only use numeric characters in SIP username entries. You cannot use a “;” *(Semicolon)* character in password entries.

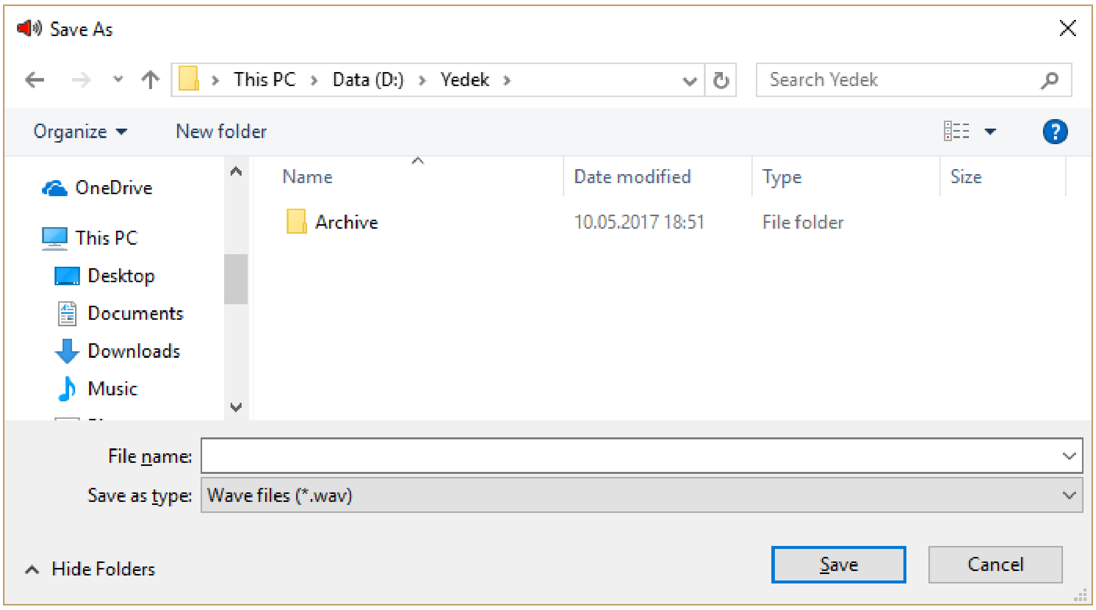
SIPob Commercial Edition

Freeware edition of SIPob does not allow play out more than 3 seconds.

How to Record a Custom Audio Message

You can use Windows Sound Recorder to record a custom audio prompt. You can use TekRecorder to record audio files compatible with SIPob.

Click record button to start recording. Click record button again after finishing. Select "File/Save As" option from File menu.



Audio file will be saved in "8000 Hz; 16 Bit; Mono" format. You can download TekRecorder from KaplanSoft website download section.

