TekIVR

Installation & Configuration Guide
Version 2.7

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http://www.kaplansoft.com/

TekIVR is built by Yasin KAPLAN

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

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Introduction

TekIVR is a SIP Interactive Voice System *(IVR)* which provides "Call Attendant" function *(Based on RFC 3261)* runs under Windows *(Vista/7/8/10, 2008-2022 Server)*. Visit https://www.kaplansoft.com/TekIVR regularly for updates.

Major features:

- Simple, easy to use interface.
- You can create your own IVR scenario using built-in scenario editor. You can select your own audio files to be used in IVR scenario *(They must be valid wav files)*.
- TekIVR supports TTS with Microsoft SAPI, Google Cloud SAPI, ElevenLabs SAPI, Speechify SAPI and MRCPv2. You can use Speech Synthesis Markup Language *(SSML)* while defining prompts.
- TekIVR supports Automatic Speech Recognition *(ASR)* with Microsoft SAPI, Google Cloud SAPI, Azure Cloud SAPI and MRCPv2.
- TekIVR can act as Proxy between MRCP v2 based application servers and SAPI-Google Speech based speech engines. TekIVR allows MRCP v2 based application servers to use SAPI and Google Speech based TTS and ASR services *(Commercial license is required)*.
- Supports G.711 A - Mu law and G.722 codecs.
- Supports NAT traversal. TekIVR also supports UPnP.
- Call transfer accomplished by using SIP REFER method *(RFC 3515)*, Bridge or by sending DTMF *(RFC 2833 / SIP INFO)*.
- TekIVR has a built-in SIP presence client and can get an extensions' online status prior to transferring a call to the extension. You can also monitor presence status of extensions in the extensions tab of TekIVR Manager.
- TekIVR can record and send incoming calls for configured SIP extensions via e-mail *(Voice Mail feature)*.
- Supports, UDP, TCP and TLS transports with RTP and SRTP.
- You can dial into TekIVR and listen recorded messages in your mailbox. TekIVR supports Voice Mail Indication *(RFC 3842)*.
- HTTP interface. All functions implemented in Win 32 GUI can be accessible through HTTP interface.
- You can monitor active SIP calls in real-time.
- TekIVR supports OpenAI real time API through Chat action in scenarios.

TekIVR uses UDP port 5072 and above for RTP traffic. You need to add necessary mappings to your router for incoming RTP traffic if TekIVR installed behind a NAT gateway which does not support UPnP.

System Requirements

TekIVR requires Microsoft .NET Framework 4.8 installed with the latest patches. Pentium i3 class CPU with 1 GB of RAM is ideal for most configurations.

Installation

Unzip “TekIVR.zip” and click “Setup.exe” comes with the distribution. Follow the instructions of setup wizard. Setup will install TekIVR Manager and TekIVR Service, add a shortcut for TekIVR Manager to desktop and the start menu.

Configuration

Run TekIVR Manager from Start Menu / Program Files / TekIVR. TekIVR automatically configures itself at first run. TekIVR selects first available IPv4 address and make a reverse lookup of that IPv4 address to obtain SIP domain information. If TekIVR cannot resolve selected IP address to an alphanumeric FQDN address, selected IPv4 address is used as SIP domain.

TekIVR also checks if it is installed behind an UPnP supported NAT gateway. If so, TekIVR automatically detects external IP and displays it on status bar. TekIVR also adds a reverse mapping for incoming UDP connections automatically *(Default UDP port 5070)*.

Settings / Parameters Tab

Click Settings / Parameters Tab to start configuration.

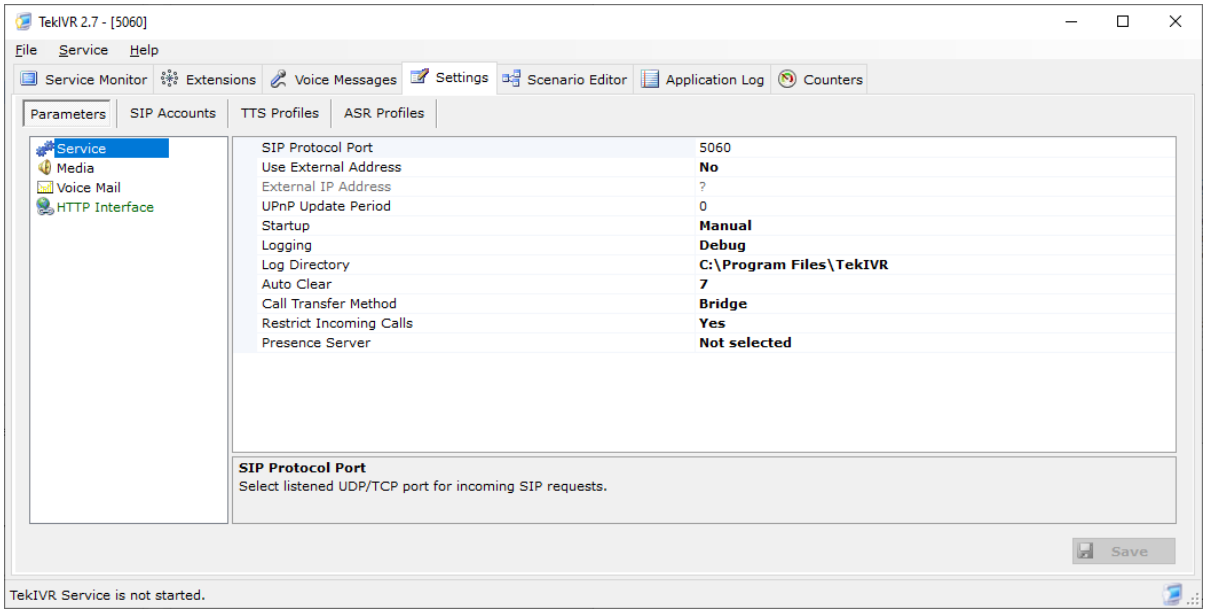


Figure - 1. TekIVR IVR Settings

Enter the following information in Service Settings

- **SIP Protocol Port:** TekIVR listens to all IP addresses on your system by default. You can define a port number to be listened to *(Default 5070)*.
- **Use External Address:** If TekIVR is installed behind a NAT gateway which does not UPnP you can set external IP address manually for NAT traversal. If your NAT gateway supports UPnP set UPnP Update Period to value greater than “0”.

- **UPnP Update Period:** You can specify the period of querying UPnP Internet Access Gateway. Set "0" to disable UPnP support.
- **Logging:** Select logging level of TekIVR. 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 <Application Directory>\Logs directory by default.
- **Restrict Calls:** TekIVR will accept calls only from its upstream routes specified in SIP accounts section if this option is set. TekIVR will accept calls from unspecified SIP endpoints if no SIP account is specified even if this option is set.
- **Presence Server:** TekIVR can track the status of defined extensions in extensions tab. You can specify which SIP presence server *[SIP server (Registrar) or IP-PBX]* holds status information for defined extensions. You can specify SIP servers in the Settings / SIP Accounts tab.

Enter the following information in Media Settings

- **DTMF Transport:** TekIVR supports receiving DTMF digits via RFC 2833 and SIP INFO requests, but you must set DTMF delivery method to remote SIP endpoints.
- **Supported Codecs.** You can choose supported codecs; G.711 A-Law, G.711 Mu-Law, G.722 and G.729.
- **Transfer Method:** Select call transfer method, either SIP REFER, Bridge or DTMF.
- **Max. Record Duration:** You can limit the maximum duration for a recording. Zero means there is no limit.
- **RTP Port Range:** Set upper and lower UDP ports for RTP port range.
- **TTS Cache Capacity:** TekIVR allows you to cache synthesized audio when this option is enabled. This reduces the number of requests made to TTS engine for identical text inputs. You can limit the number of cached items to be kept in memory. Specify a value greater than zero to enable TTS caching.
- **MRCP Proxy:** Click to enable MRCP proxy.

Enter the following information in Voice Mail Settings

- **Welcome Message:** You can optionally select your own recorded welcome message which will be played when incoming calls arrive. File format must be 16 bit per sample, 8 KHz and 1 channel.
- **Use TLS Transport:** Enable TLS transport to connect to the SMTP server.
- **Ignore TLS Errors:** Ignore certificate related errors and warnings.

- **SMTP Server Address:** Enter IP address or FQDN of the SMTP server.[1]
- **SMTP Server Port:** Enter TCP port number for the SMTP server.
- **Mail From:** Enter e-mail address which will be shown as sender address.
- **Display Name:** Enter sender display name.
- **Authentication Required:** Check it if your SMTP server requires user authentication.
- **SMTP Username:** Enter SMTP username.
- **Password:** Enter password of SMTP user.

Enter the following information in HTTP Interface Settings

- **Enabled:** Enable HTTP interface.
- **HTTP Port:** Enter TCP port number for the internal HTTP server.
- **HTTP Password:** Enter the password of the HTTP interface.

Settings / SIP Accounts Tab

Click Settings / SIP Accounts settings.

Type a SIP Account name to the bottom left text box and click the add button right to it. Enter the following information for a SIP account.

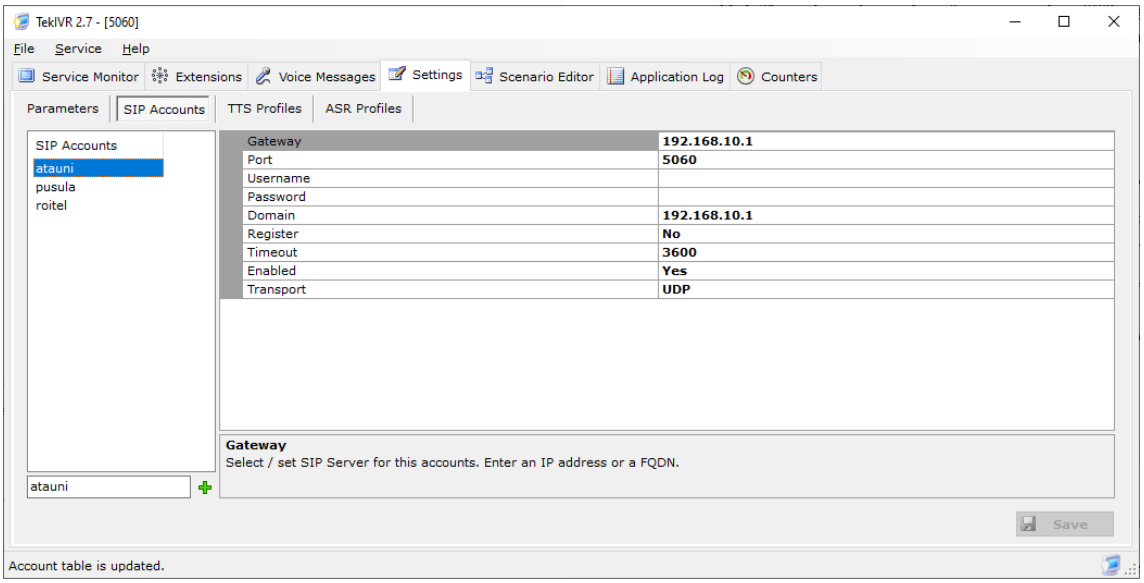


Figure - 2. TekIVR SIP Accounts

- **Gateway:** Enter an IP address or FQDN of your SIP gateway or proxy. Please make sure that this address is resolvable by your SIP client and has a valid entry *(an A record)* in your DNS server if you enter FQDN. You must configure a valid SIP proxy for proper operation.

[1] Please set SMTP port as 587, go to https://myaccount.google.com/lesssecureapps and set Allow less secure apps: ON if you prefer to use an Gmail account.

- **Port:** Enter signaling port for the SIP server. The default value is 5060.
- **Username:** You must specify a SIP endpoint ID for TekConSer. If incoming calls are not destined to the number specified in Account ID parameter, TekConSer will reject the calls. This ID is also used by TekConSer while registering to SIP registrar.
- **Password:** If SIP registrar requires password authentication specify password.
- **Domain:** Enter a FQDN or an IP address of your SIP domain. You must configure a valid SIP domain for proper operation.
- **Register:** SIP endpoint registration is disabled by default. If this option is enabled TekConSer tries to register itself to configure SIP Domain.
- **Timeout:** Set registration timeout for this SIP account. TekConSer will re-register after timeout.
- **Enabled:** Select "Yes" to enable this SIP account.
- **Transport:** Select transport for this SIP server *(TCP or UDP)*.

Registered SIP account will be displayed in **green** color. You can delete a SIP account by pressing the delete key on keyboard after selecting.

If you click [Save] button setting will be saved and activated immediately.

Settings / TTS Profiles Tab

TekIVR allows you to create multiple TTS profiles to be used in text to speech operations. You can specify a locale, speech engine and voice type for each TTS profile;

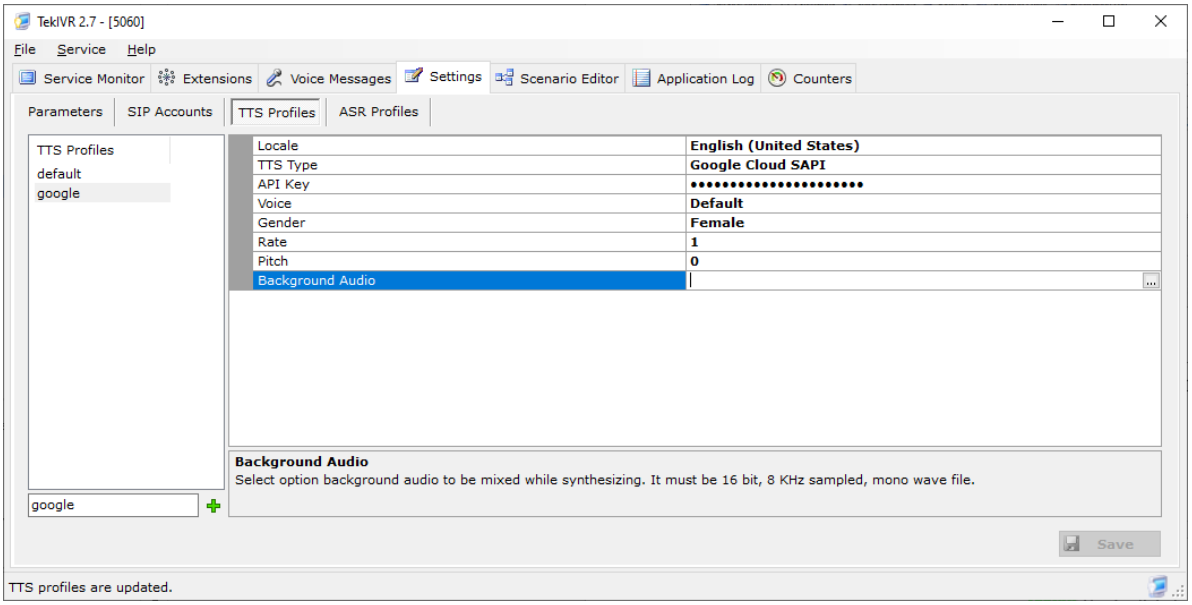


Figure - 4. TekIVR TTS Profiles

You must configure a valid Azure, Google or ElevenLabs API key prior to using Azure, Google Cloud, ElevenLabs SAPI or Speechify SAPI as a TTS option. You cannot list Voice options for Azure and Google Cloud SAPI without having a valid API key.

Settings / ASR Profiles Tab

TekIVR allows you to create multiple ASR profiles to be used in speech recognition operations. You can specify a locale and speech engine for each ASR profile;

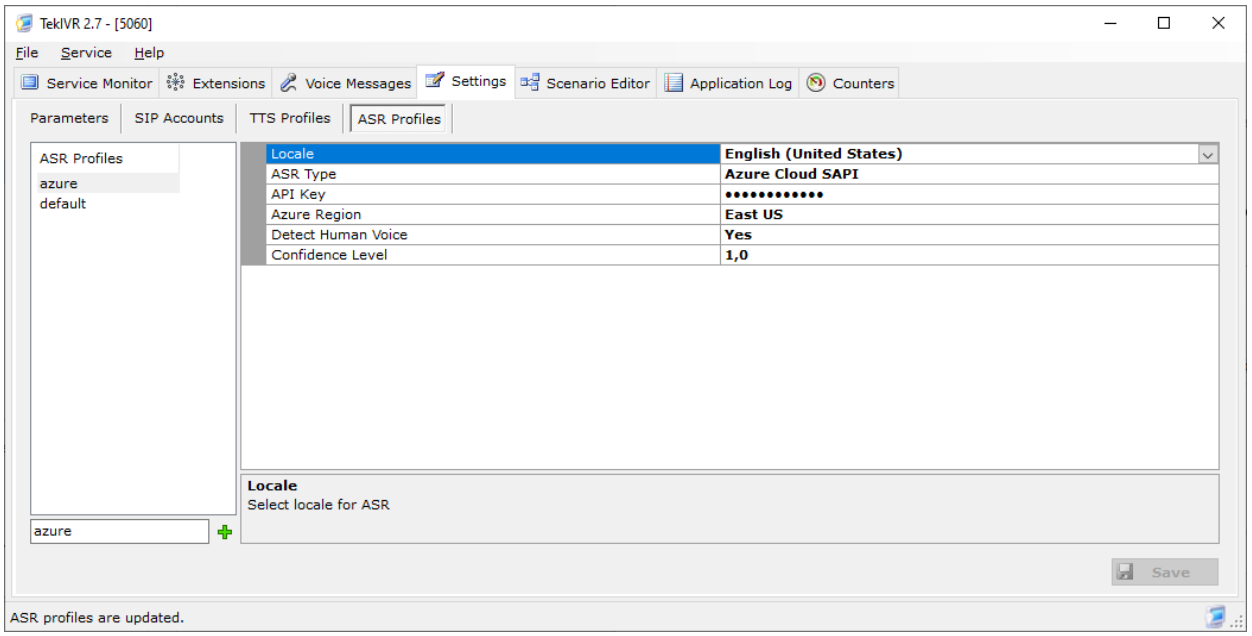
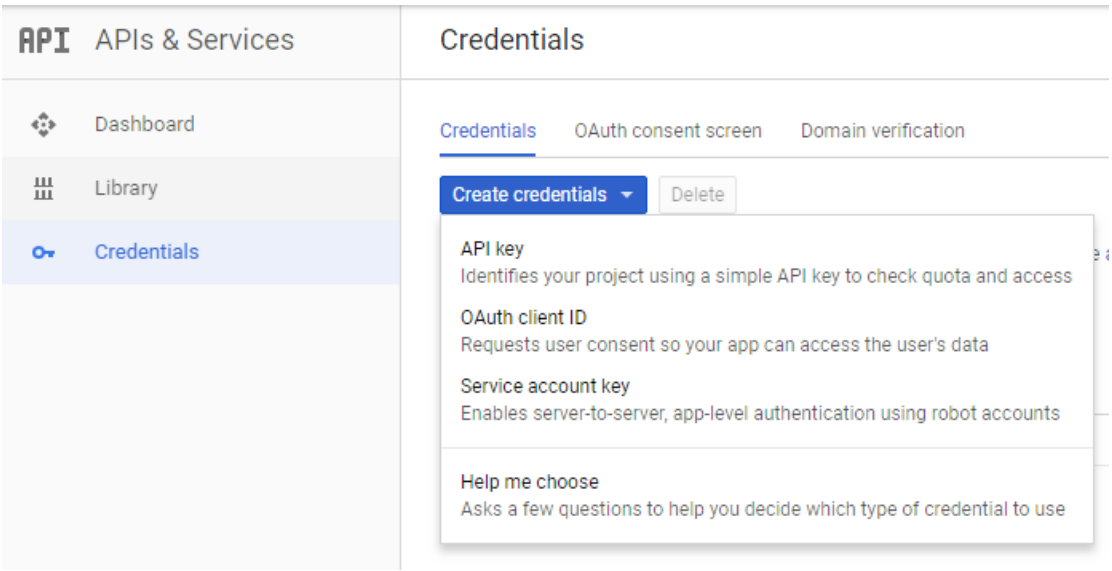


Figure - 5. TekIVR ASR Profiles

Google API Key

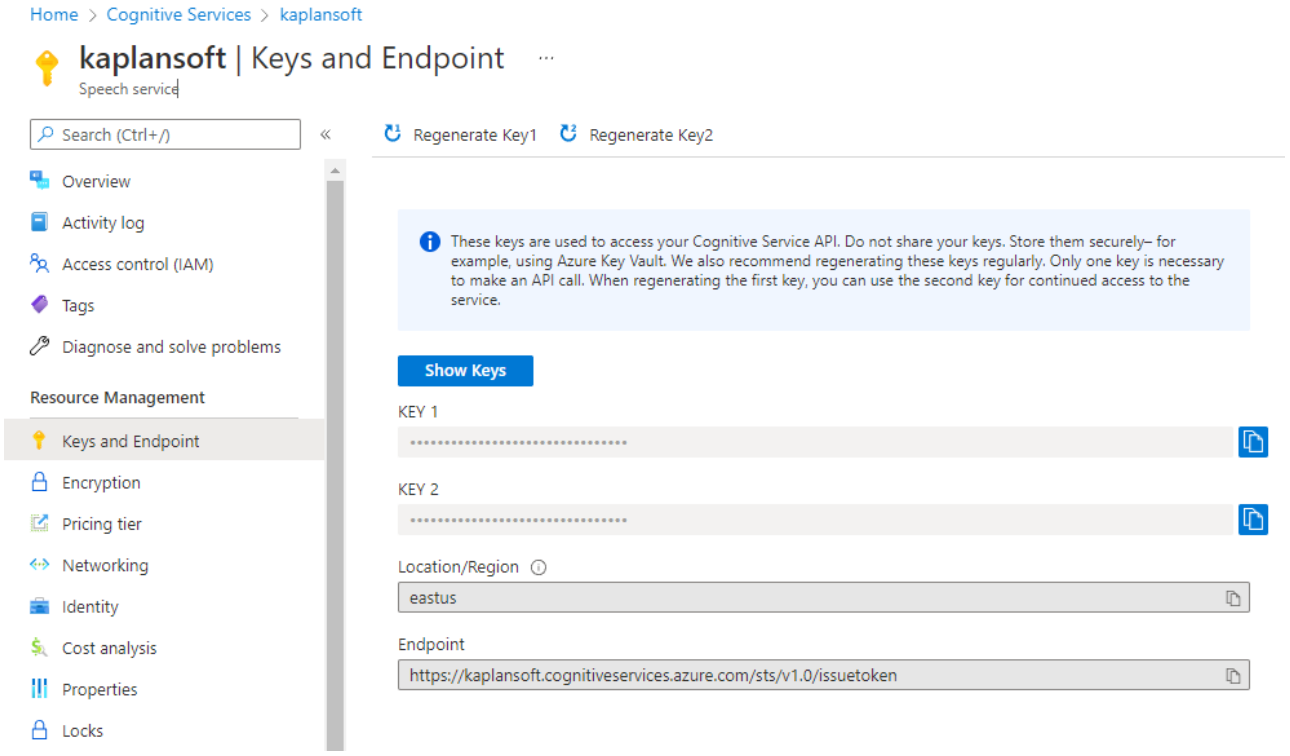
You need to have a Google API key created for Cloud Speech-to-Text API and/or Text-to-Speech API. To create API key, go to Google Cloud Platform and create a project first.

1. Navigate to the APIs & Services -> Credentials panel in Cloud Platform Console.
2. Select Create credentials, then select API key from the dropdown menu.
3. Click the Create button. A dialog box appears which displays your newly created key.



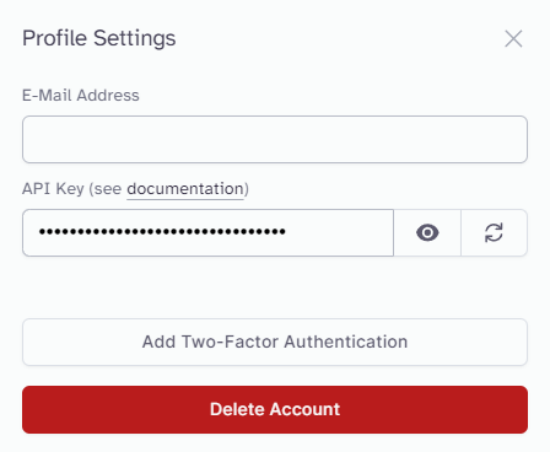
Azure API Key

You need to have an Azure API key created for Cloud Speech-to-Text API and/or Text-to-Speech API. To create an API key, go to Azure Portal, create a service under Cognitive Services. You can find your keys under Keys and Endpoint. You can use the first key. You can set your region while creating the service entry.



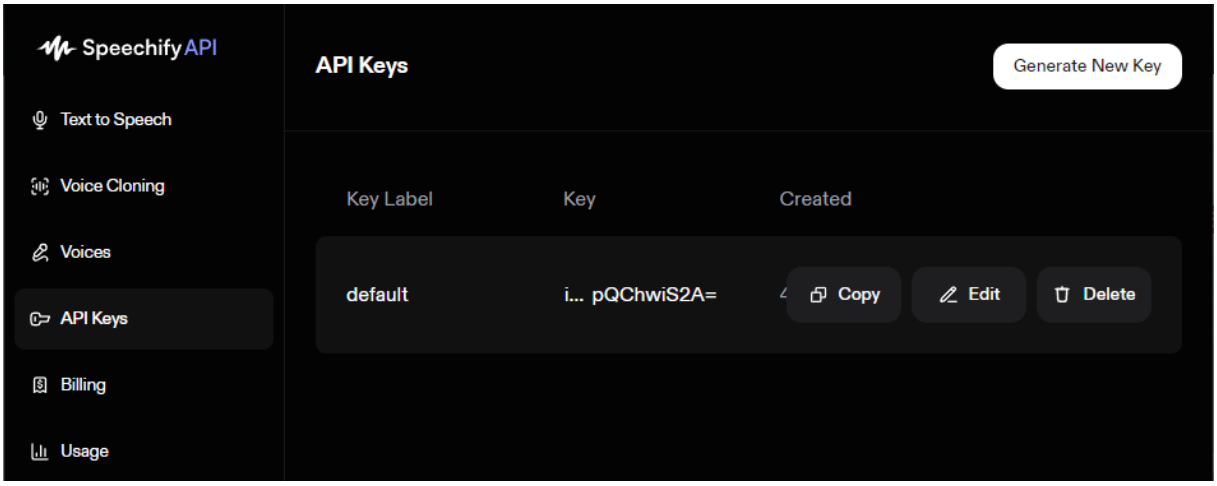
ElevenLabs API Key

You need to have an ElevenLabs key created for Text-to-Speech API. Your API key is accessible through your account profile settings.



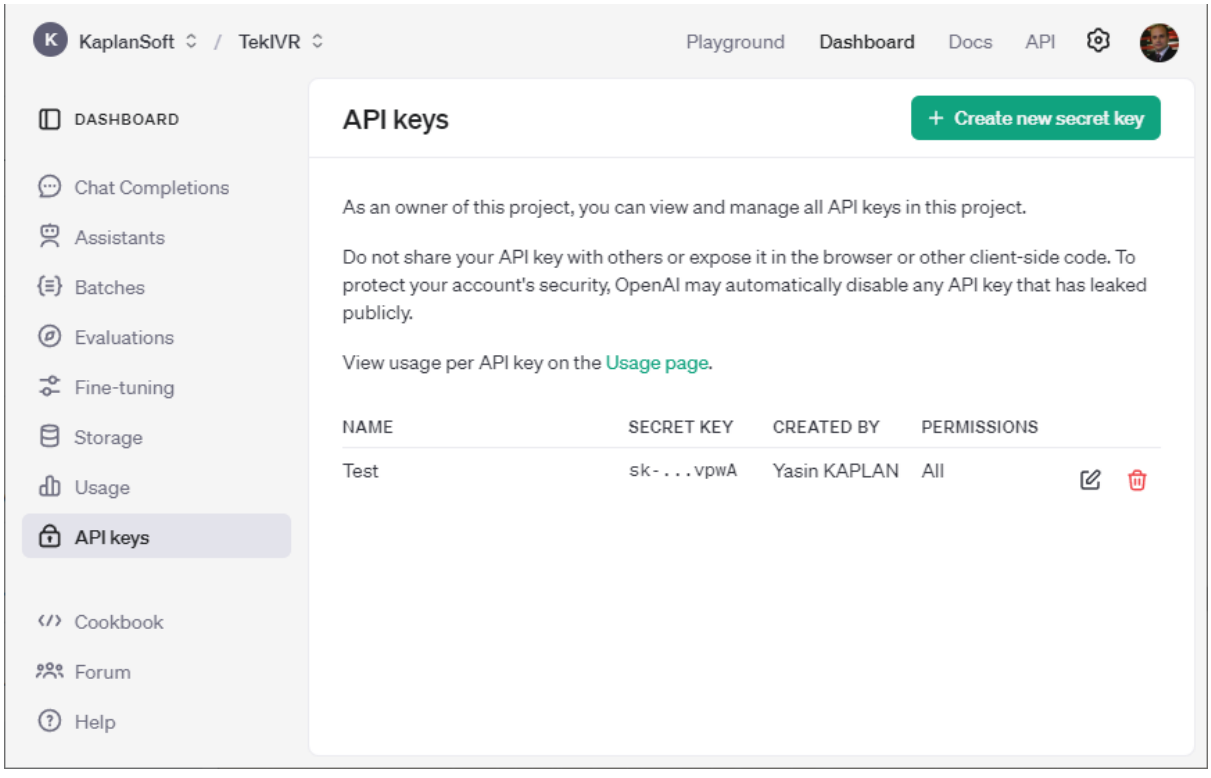
Speechify API Key

You need to have a Speechify API key. Your API key is accessible through Speechify console.



OpenAI API Key

You need to have an OpenAI key created for real time API. Your API key is accessible through Organizations / Projects / API Keys section.



You can enter port number with IP address for the MRCP Server parameter in <IP Address>:<Port> format *(192.168.10.10:5060 e.g.)*. Default port 5060 will be used when the port number is omitted. Default transport is TCP for MRCP connections. You specify UDP transport by adding `;transport=udp` suffix to the server address *(192.168.10.10:5060;transport=udp e.g.)*.

Service Monitor

You can monitor active calls through the service monitor tab. You can clear log entries by right clicking on Call Log.

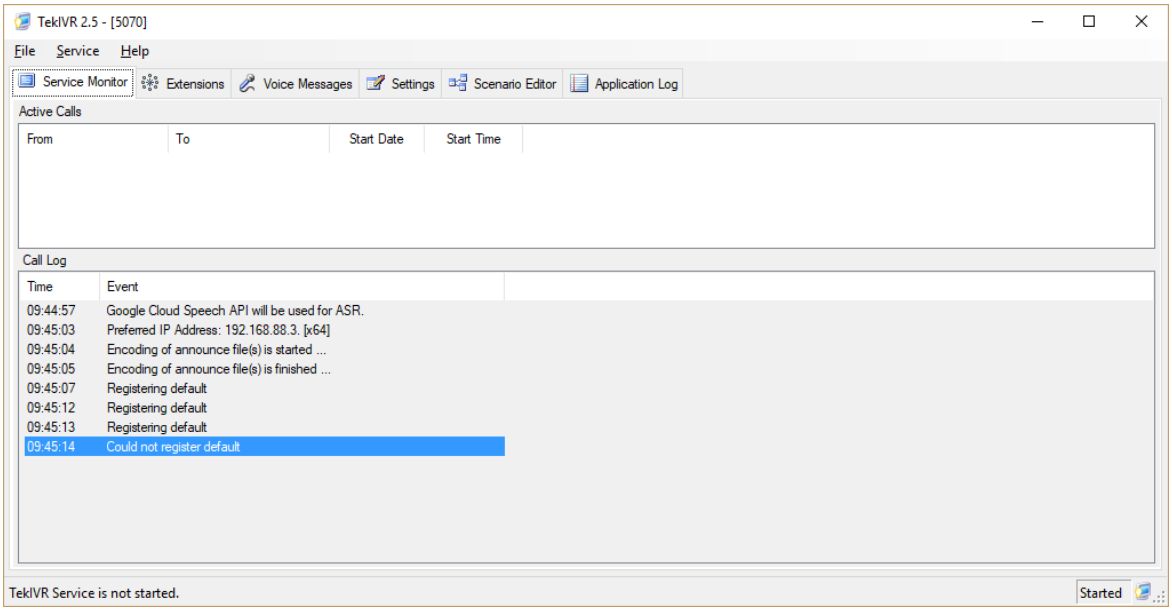


Figure - 6. Service Monitor Tab

Extensions

You can define extensions, their mailbox numbers and optionally their e-mail address for sending recorded audio messages through Extensions tab. You can also monitor the status of extensions if you have specified a presence server in Settings / Service.

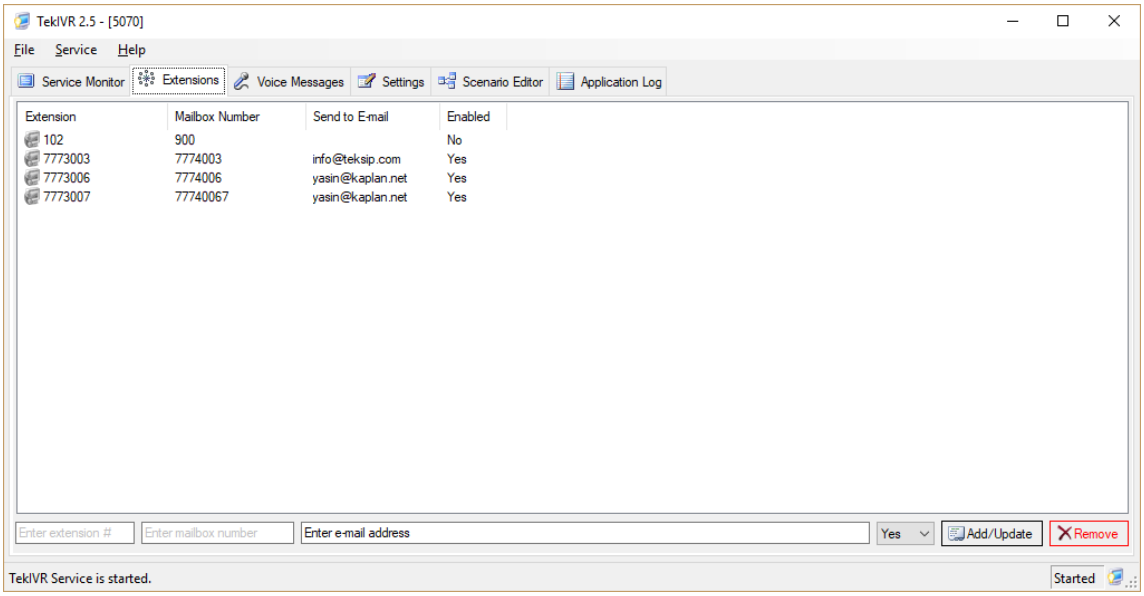


Figure - 7. Extensions Tab

Scenario Editor

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

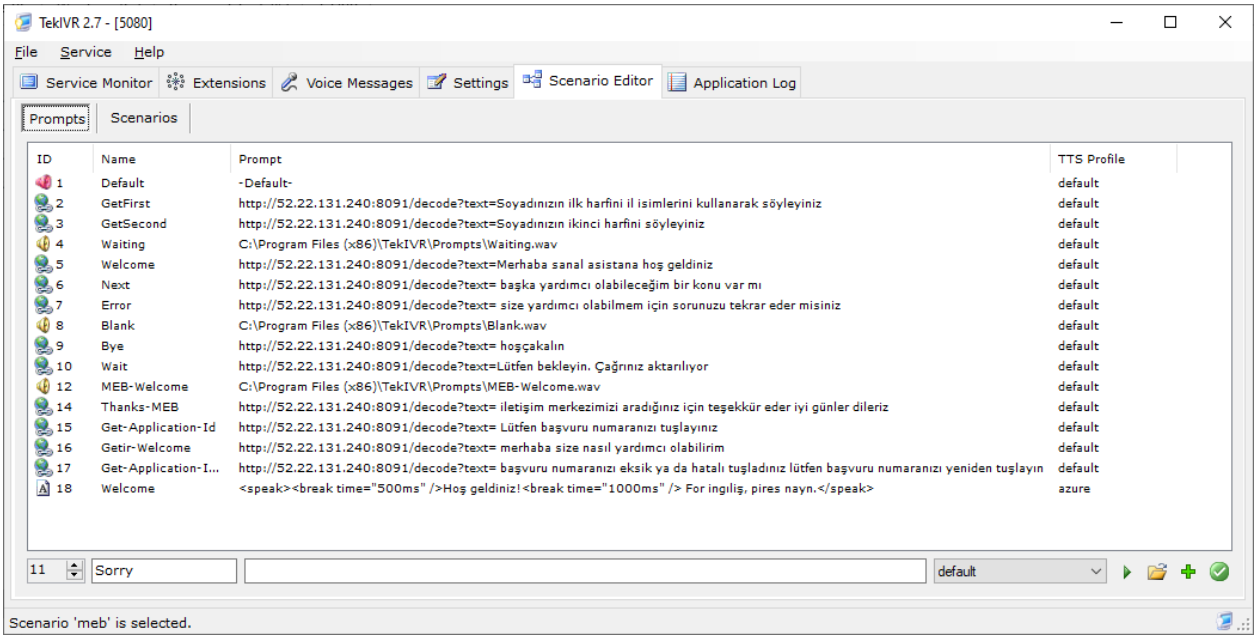


Figure - 8. Scenario Editor / Prompts Tab

You can specify a TTS profile if you enter a text to be synthetized. TTS profiles can be created at Settings / TTS Profiles 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 have a descriptive name for the prompt. "-Default-" specifies built-in welcome announce.

You can enter URLs *(http or https)* which retrieve wave resources from a web server as prompts. *(http://example.com/wavefile.wav e.g.)*

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. You can use Speech Synthesis Markup Language while defining prompts. Please see http://msdn.microsoft.com/en-us/library/hh361578(v=office.14).aspx as a reference. Sample Prompt entry;

```
<speak version="1.0" xmlns:ssml="http://www.w3.org/2001/10/synthesis" xml:lang="en-
US">Please enter pick up time 24 hours format; 2 digits for hour and 2 digits for
minutes.<break time="1000ms" /></speak>
```

Do not use speak, voice and prosody tags if you use Azure Cloud SAPI. TekIVR adds these headers automatically.

Single quote character needs to be escaped using a backslash while entering prompts for Google TTS. TekIVR removes the escape character automatically prior to submitting the same text to other TTS engines.

```
<speak>Please enter pick up time 24 hours format; 2 digits for hour and 2 digits for
minutes.<break time=\"1000ms\" /></speak>
```

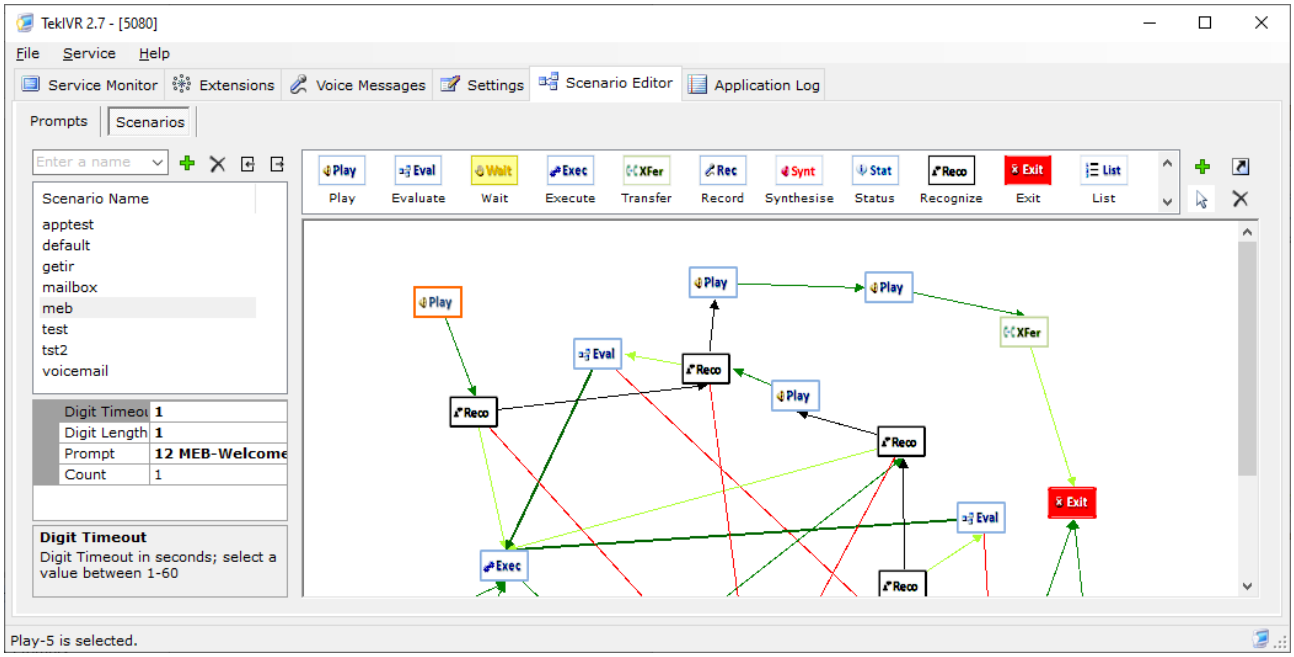


Figure - 9. Scenario Editor / Scenarios Tab

Scenario Editor / Scenarios Tab

Click Scenario / Scenarios Tab to define scenarios and steps in your IVR 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 IVR scenario is invoked by TekIVR. You should also specify the maximum number of steps that can be executed in a scenario in order to prevent infinite loops. You can specify the entry action which is executed as the first action in a scenario. You can set the entry action by double clicking on an action icon in scenario map. 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. Termination Action allows you to run an external application, access a URL or execute an SQL statement through ODBC. Syntax is like the Execute action described below but path and parameters entered together unlike Execute action. Use `ODBC:DSN Name:SQL Statement` format for SQL statements. TekIVR allows some internal and custom variables to be used in action parameters. Variable names are case sensitive.

- `%recognized%` The output of the last Recognize action used in the scenario.
- `%pasttext%` The output of the previous Recognize action used in the scenario.
- `%execout%` The output of the last Execute action used in the scenario.
- `%pastresponse%` The output of the previous Execute action used in the scenario.
- `%received_digits%` The output of the previous Execute action used in the scenario.
- `%called_number%` The number which is found in To: header.
- `%calling_number%` The number which is found in From: header.
- `%sipdomain%` SIP domain for the calls.
- `%date%` Current date and time in current locale format.
- `%saved%` Full path of the latest recorded audio file by the Record action.
- `%callid%` SIP Call-Id of the call.
- `%sessionid%` Locally generated unique session if for the call.
- `%custom%` The custom parameter can be defined for the outgoing calls.
- `%telnum%` The number which is found in To: header.

- `%custom_variable_0%` User defined custom variable.
- `%custom_variable_1%` User defined custom variable.
- `%custom_variable_2%` User defined custom variable.

TekIVR supports the following actions in IVR scenarios:

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 the next step after the action is completed *(NextAction)*.

Synthesize. This action is like Play action and has the same parameters. However, Synt synthesizes audio prompt dynamically using selected TTS engine by processing test string specified in Prompt parameter. You can use internal variables in prompt text. You can also specify a URL *(http/https)* to fetch text from a web server. If the fetched resource is a wave file, it will be played out directly otherwise if it is a text file it will be used as input for the TTS engine. You cannot use pipe "|" character in a prompt text.

Record. Record action will record audio from the calling part after playing out a prompt which is configurable. You can restrict the maximum duration of the audio recording by the Duration parameter. Set it to 0 for unlimited duration. Caller can terminate recording by dialing pound key *(#)*. You must also define the next step after the action is completed *(NextAction)*.

TekIVR will record audio from the caller into memory if the file path is set to blank. TekIVR will create sub directories for each unique called number under specified directory and save audio files under these sub directories created for called numbers. Audio files named as <Calling Number>-YearMonthDayHHmmss-ms.wav where the date is file creation date *(2123561212\9990000-20201215130015-430.wav e.g.)*.

You can play out recorded audio to the called party prior to transferring an incoming call to an internal extension. This can be used to identify callers by their own voice to called parties. Please also see Transfer action for other details.

Wait. You can force the caller to wait while playing out a configurable prompt. You can specify wait time in seconds. Set Detect Human Voice = 'Yes if you would like to detect human voice. Next action will be performed upon human voice detection. You must also define the next step after the action is completed *(NextAction)*.

Evaluate. Evaluate action evaluates user input and then determines the next action. If TekIVR cannot find an action that 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. The Execute action executes the executable defined in *ExecutablePath* parameter. You must enter executable with full path. You can use internal variables in command line parameters *(ExecutableParameters)*. You can also pass header values from the initial INVITE message as parameters in a SIP request. Use `%header-name%` syntax *(`%call-id%` or `%user-agent%` e.g.)*. Leave *ExecutableParameters* blank if no parameter will be used. Set WaitforCompletion > 0 in seconds if you wish TekIVR to wait completion of the execution of the executable and TekIVR will

use return the value *(DOS Errorlevel)* of the executable as the next action be executed. You must also define the next step after the action is completed *(NextAction)*. If you set WaitforCompletion > 0 TekIVR 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, TekIVR will connect to the URL, get response and will set `%execout%` variable to the web server response. You can also specify an HTTP/HTTPS URL in *ExecutablePath* parameter. TekIVR will perform an HTTP GET action if there is not any parameter otherwise TekIVR will perform an HTTP POST action.

Executable Path	**http://www.kaplansof.com/MyAppp?Number=%called-number%**
Executable Parameters	
Wait for completion	**60**
Encoding	**ASCII**

The Execute action allows you to perform a database query through an ODBC source. You must create an ODBC connection profile (64 Bits) in system DSNs. Here is an example;

Executable Path	**ODBC:MyDataSource**
Executable Parameters	**Select Label from MyTable where Number = '%called_number%'**
Wait for completion	**60**
Encoding	**ASCII**

You can set the next action based on the query result in the executable parameters. You can specify the next action based on failure and success cases. TekIVR chooses failure action if query returns nothing or returned scalar value equals or less than zero. You can specify success and failure action using the following notation;

```
ODBC:<Data source name>:<Success action id>:<Failure action id>
```

TekIVR can connect to a UDP, TCP or TLS endpoint and send/receive data if you specify a destination in the Executable Path parameter

```
[udp|tcp|tls]://kaplansoft.com:600
```

You can set `%received_digits%`, `%display_name%`, `%sessionid%`, `%execout%`, `%custom_variable_0%`, `%custom_variable_1%` and `%custom_variable_2%` variable values from the output of the executable. For instance, you can set `%received_digits%` by configuring your executable to return

```
received_digits=desiredvalue
```

TekIVR will parse returned output and set `%received_digits%` to *`desiredvalue`*. You can set multiple variables

```
received_digits=desiredvalue;custom_variable_0=desiredvalue
```

You can also directly set internal variables `%received_digits%`, `%display_name%`, `%sessionid%`, `%execout%`, `%custom_variable_0%`, `%custom_variable_1%` and `%custom_variable_2%` using Execute action. Enter `set <variable name>` to Executable Path and enter desired value to Executable Parameters:

Executable Path	**set execout**
Executable Parameters	**my value**
Wait for completion	**10**
Encoding	**ASCII**

Transfer. Transfer action will transfer the call to the extension defined in *Number* parameter. You can have a predefined extension or internal 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 the next action for failed and successful call transfers.

"Play Transferee Prompt" option enables you to play out an informative message prior to transfer the call. You can also record an audio message from the caller party and play out the called part. You must have a Record action in your scenario with a blank file path to have a recorded audio message from the called party. TekIVR will play out caller audio message and then selected "Transferee Prompt". Called party can accept the call by pressing 1 or repeat the message by pressing 2. Pressing 3 will reject the incoming call. Call will be transferred if the called party does not dial any DTMF digit. A sample "Transferee Prompt" would be "… is calling you. Press 1 to accept the call, press 2 to repeat the message or press 3 to reject this incoming call. Stay on the line if you would like to accept the incoming call"

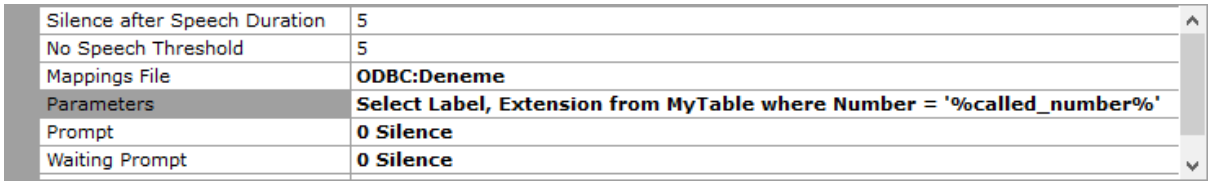
The transfer function allows you create a hunt group. You can specify extensions in the hunt group, by entering multiple extension numbers concatenated by commas. Please see "Creating Hunt Groups in Scenarios" section for more details.

Recognize. TekIVR can recognize user input via MRCP supported ASR system, Google/Azure Cloud Speech API and Microsoft Speech Recognition engine. You can create an ASR profile for each Recognize action. You need to specify a prompt and a mapping file as parameters for this action. Mappings file is a CSV file containing audio input and corresponding values. Recognized input will be stored in `%recognized%` and if recognized text has a corresponding mappings entry, mapped value will be stored in `%received_digits%` variable. Sample entries for Mappings file;

```
"John Doe","101"
"Jane Doe","102"
"Yasin KAPLAN","102"
```

TekIVR will execute and process console output if you specify a mapping file with .exe, .bat or .vbs extension. Please note that console output must be identical to a mapping file content as shown above. You can also specify parameters for the executable. Parameters internal variables.

You can also specify an ODBC data source to fetch mapping entries. An example is shown below;



Data tables must have at least two fields; one for item to be recognized and the other for corresponding value *(Number which will be used for call transfer e.g.)*. Your query must return these fields in this specified order.

You can have three destination actions for Recognize action; one for successful recognition, second is for failed recognition and third for no user input *(Total silence)*. Success action is set in the first link. Create a second link from Recognize action for failure action and a third link for no input action.

TekIVR can detect silence. TekIVR will start recognition after silence duration after speech. The default silence duration is 2 seconds. You can increase or decrease for better results. You can also set recognition timeout. TekIVR will choose failure action if recognition timeout expires.

Status. Status action checks presence status of an extension and select defined action based on presence status. You can specify the next action for online, offline and busy status. You can specify a fixed number or use internal variables. Variables are case sensitive.

List. List action can be used to play out audio files in a directory. Audio files must be in 16 bits, 8KHz, mono format. List action returns the number of audio files found in specified directory. This value can be evaluated by the Evaluate function. List action allows you to specify info prompts to be used with this action. List accepts prompt text and synthesizes them using specified TTS profile.

Send. Send specified digits to the called party. You can use internal variables and specify inter digit delay in milliseconds.

Dial. TekIVR allows you to dial a specified SIP destination and execute a specified child scenario when the remote party answers. Collected digits and executable output are returned to the parent scenario. TekIVR can connect the calling party of the parent scenario to the called party of the child scenario when you terminate child scenario with an Exit action with Jump to Scenario = Bridge for an audio conversation session.

Prompt	**0 Silence**
Timeout	**10**
Execute Scenario	**auxilary**
Dial Number	**102**
Use Account	**teksip**

You can also use internal variables for the number parameter. TekIVR sends a re-INVTE to the calling party to change RTP endpoint information in Bridge method in the call transfers. This may not be possible when the call is received from a cloud telephony provider. You can use dial action as an alternative call transfer method in such cases.

Exit. Exit action terminates execution of IVR 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 TekIVR will also terminate execution of IVR scenario.

Exit action enables you to jump to an existing scenario. You can choose a scenario to be jumped by setting “Jump to Scenario” option. TekIVR can connect the calling party of the parent scenario to the called party of the child scenario *(Invoked when a Dial action is executed)* when you terminate child scenario with an Exit action settings Jump to Scenario = Bridge for an audio conversation session.

Chat. TekIVR Chat action in scenarios allows you access to OpenAI real time API. OpenAI real time API allows you to interact with OpenAI via voice. OpenAI real time API also allows you to execute user define functions and get / post data to enterprise databases and communicate with enterprise applications. Chat action has the following parameters:

Chat Server URI	wss://api.openai.com/v1/realtime?mode
API Key	sk-proj-0yEt7aQwu3pJ777777E3d7_QfgCF
Voice	ash
Maximum Response Output Tokens	0
Instructions	You are a helpful call receptionist for con
Functions	F:\Operations\Tools.json
Function Mappings	F:\Operations\Tools.csv

- **Chat Server URI.** This is a secure WebSocket URL *(WSS)* to connect (wss://api.openai.com/v1/realtime? model=gpt-4o-realtime-preview-2024-10-01)
- **API Key.** This is obtained from OpenAI dashboard Organizations / Projects / API Keys section. You need to create a project for the IVR scenario.
- **Maximum Response Output.** The maximum number of tokens that can be generated in the chat completion.
- **Instructions.** The system instructions that the assistant uses. Instructions describe overall assistant behavior. You should restrict assistant operation only for transferring incoming calls to the desired department or extension. Example:

  You are a helpful call receptionist for company called KaplanSoft and you always speak in English. Greet the calling party with saying \"Welcome to KaplanSoft! How may I help you?\" without waiting for user input. Customer can ask his or her order status and customer can also request his or her call to be transferred to a company department.

  You can enter a full path of text file *(C:\Program Files\TekIVR\Prompts.txt e.g.)* containing instructions if you have a multi-line instruction.
- **Functions.** Full path of the JSON formatted text file which contains user defined functions which can be used by the assistant. Descriptions can be written in your native language if it's supported by OpenAI. Try to be precise and use minimum words as much as possible.

  There are two built-in functions in TekIVR; transfer_call and terminate_call. You can override them by specifying your own functions. Default definitions for these built-in functions:

```
[
  {
    "type": "function",
    "name": "transfer_call",
    "description": "Transfer call to an extension",
    "parameters": {
      "type": "object",
      "properties": {
        "extension": {
          "type": "string",
          "description": "Extension number of the desired extension"
        }
      },
      "additionalProperties": false,
      "required": [
        "extension"
      ]
    }
  },
  {
    "type": "function",
    "name": "terminate_call",
    "description": "Terminate the current conversation. This function is called when the calling
                    party has no further requests or says bye."
  }
]
```

transfer_call function sets TekIVR internal variable `%received_digits%` to received extension value. You need to evaluate `%received_digits%` variable value using Evaluate action and perform Transfer action based on Evaluate action. This typically done connecting Chat action to Evaluate action and connecting Evaluate action to Transfer actions.

- **Function Mappings.** Full path of the CSV file which contains function to local executable or web service URL mappings.

  You can perform a HTTP GET request to a web server URL or invoke a DOS batch file prior to transferring a call for instance. You can use TekIVR internal variables `%called_number%`, `%calling_number%`, `%recognized%`, `%received_digits%` *(User input)*, `%date%`, `%execout%` and `%sipdomain%` or returned parameters for the function defined in parameters section. The parameter returned for transfer_call function is "extension". Example mapping file:

```
transfer_call,"F:\Operations\cdr.bat %calling_number% %called_number% %extension%"
```

  Contents of cdr.bat

```
@echo off
echo %1 %2 %3 >> F:\Operations\cdr.csv
echo Say \"Stay on the line, I'm transferring your call\" in the conversation language.
```

  Batch file output will be used as a function description. You can specify your own functions.

```
{
    "type": "function",
    "name": "query_order",
    "description": "Queries status of an customer order. Ask for six digits order
                    identification number if a customer asks for his or her order
                    status and query it by calling this function.",
    "parameters": {
      "type": "object",
      "properties": {
        "orderid": {
          "type": "string",
          "description": "Numerical identification number for an order"
        }
      },
      "additionalProperties": false,
      "required": [
        "orderid"
      ]
    }
}
```

  This function named query_order will return a parameter named ordered and it can be used as a parameter for a local application as shown below:

```
Qqery_order,"http://example.com/getorderstatus?orderid=%orderid%"
```

  TekIVR will assign the latest transcription text value returned from the server to the `%execout%` internal variable.

It is wise to start your IVR 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.

MRCP v2 Proxy

You can use TekIVR as a proxy for MRCP v2 based application servers to SAPI or Google Speech based TTS and ASR services. You just need to specify TekIVR as MRCP v2 server in your applications server's configuration. TekIVR currently supports only TCP transport for MRCP v2. You can deploy TekSIP as TLS to TCP transport proxy if TLS transport is required. MRCP v2 proxy feature requires a commercial license.

Custom Voice Mail and Mailbox Prompts

You can customize IVR scenarios used for Voice Mail navigation and Mailbox audio recording. TekIVR uses a scenario consisting of a Record and Exit action for voice mail. You can overwrite default voice mail scenario by creating a scenario named "voicemail". TekIVR will automatically add scenario action when you add a scenario named "voicemail".

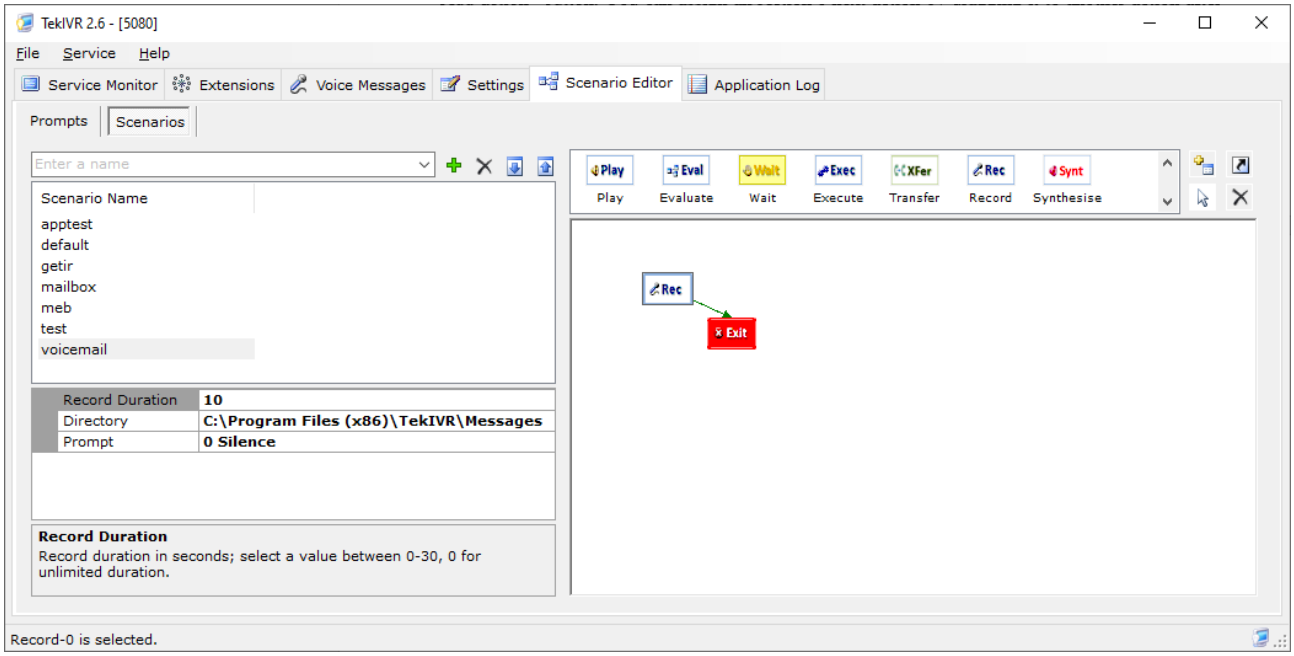


Figure - 10. Custom Voice Mail Scenario

You can change record notification prompt, record duration and the directory which audio recording files are kept. You must also create a custom mailbox scenario if you change the audio recording directory.

Add a scenario named "mailbox" if you would like to change the scenario to be executed when a user dials in to TekIVR to listen audio messages in his/her mailbox. TekIVR uses TTS to synthesize

audio prompts to be played for mailbox navigation. You can create a TTS profile for a proper language to be used for TTS.

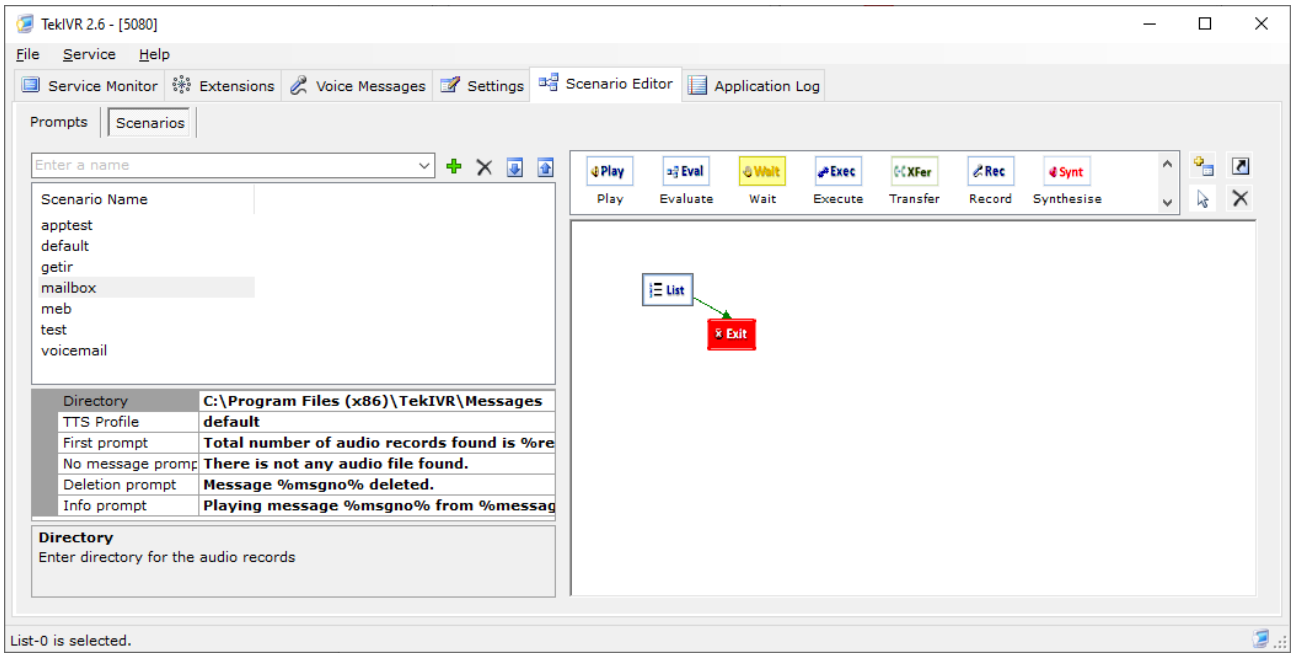


Figure - 11. Custom Mailbox Scenario

Call Flow for the Default IVR Scenario

You can see how incoming calls are processed by TekIVR's default scenario. When an incoming call arrives, TekIVR plays welcome message *(Default built-in message played if a custom is not configured)*. TekIVR starts digit collection as soon as starts playing welcome message. Digits collected processing RTP Event packets *(RFC 2833)*. TekIVR terminates digit collection and starts call transfer sequence immediately if # *(Pound)* key is detected. If the collected number of digits is less than specified number in TekIVR settings, TekIVR waits for digit timeout value. If collected number of digits still less than specified number in TekIVR settings, TekIVR re-plays welcome message. This sequence is repeated, until loop count equals value of Repeat parameter. Finally, call transfer is initiated to the phone number formed by collected digits using REFER, Bridge or DTMF method.

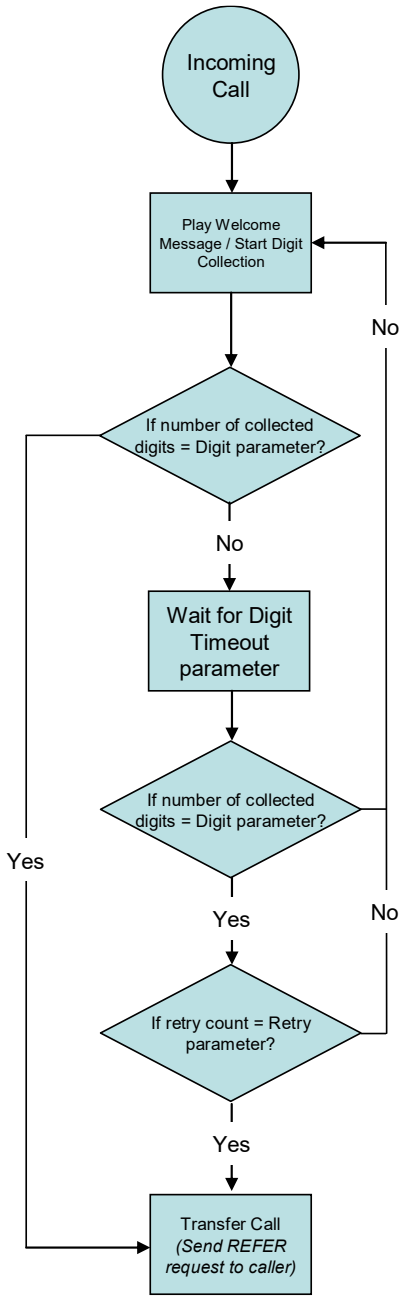


Figure - 12. TekIVR call flow

Typical Application

You can see a typical layout a sample office VoIP network consisting of a SIP Proxy/Registrar *(TekSIP e.g.)*, a VoIP gateway which provides PSTN interconnection, TekIVR system *(Can be run on the same server with TekSIP)* and IP phones. It's assumed that all units are registered to SIP registrar.

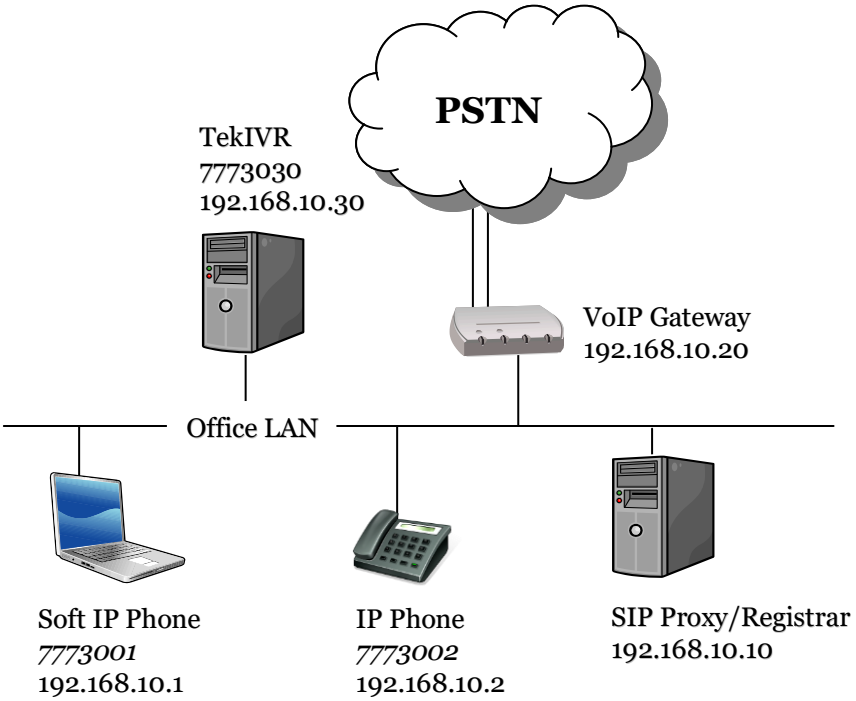


Figure - 13. Sample Network

TekIVR has the following configuration for this network:

Listen IP Address | Port : 192.168.10.30 | 5070

SIP Account entry;

SIP Domain / Gateway : 192.168.10.10
Register : Checked
Account ID : 7773030
Password : <Optional>

Other settings can be left as default. You can use any SIP VoIP gateway which supports SIP REFER or DTMF method for call transfer. You must route incoming calls from PSTN *(FXO ports if gateway connected to PSTN through analog lines)* to 7773030 at VoIP gateway. Most VoIP gateways provide this feature. Look for port settings to find out how to forward incoming calls to a default number. VoIP gateway must be configured to register to the SIP registrar. Use the same domain configured for the SIP account *(192.168.10.10 in our example)*.

You can have optionally other kinds of VoIP devices on your network such as VoIP soft phones.

When an incoming call arrives to the VoIP gateway, the VoIP gateway will initiate a call to 7773030 through SIP Proxy. You can see a SIP ladder diagram showing SIP messages exchanged between the SIP endpoints:

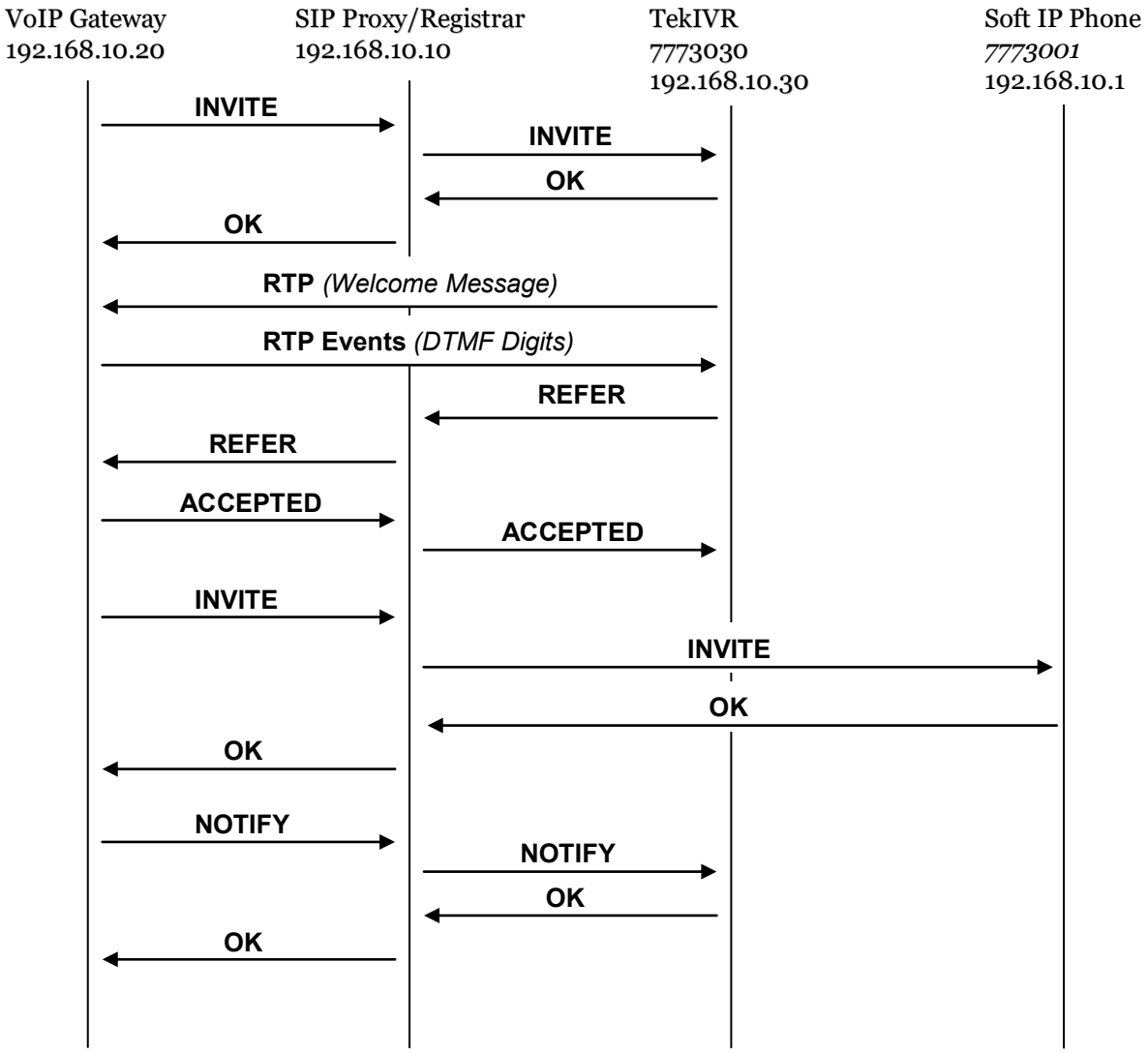


Figure - 14. Exchanged SIP Messages during Call Transfer

TekIVR stays in signaling path when using Bridge method for call transfers. TekIVR establishes a SIP dialog with destination number and then redirect RTP path sending by a re-INVITE to the calling party.

You can dial into TekIVR to listen recorded messages in your voice-mailbox. Dial your Mailbox number to listen recorded messages.

Troubleshooting

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

Recorded messages saved under Messages subdirectory of TekIVR application directory.

TekIVR Messages

Settings could not be loaded. Initializing with default values.
TekIVR Service is being started with default values.

You get this message at the first run of TekIVR. If TekIVR cannot find or read TekIVR.ini initializes itself with default settings.

Unable to initialize UDP/TCP thread [5060]

If another application is configured to use same UDP/TCP port with TekIVR, TekIVR cannot initialize respective thread.

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.

How to Record a Custom Welcome Message

You can use TekRecorder to record audio files compatible with TekIVR.

Figure - 15. TekRecorder

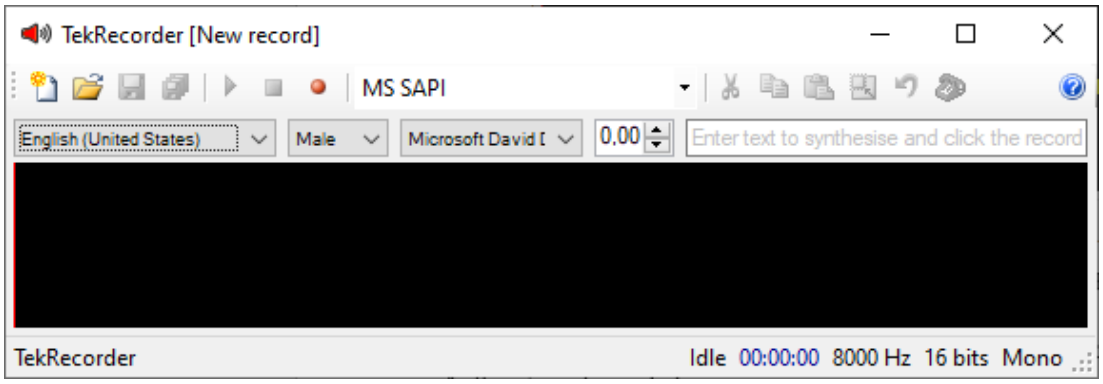


Figure - 16. TekRecorder can synthesize audio using MS SAPI

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

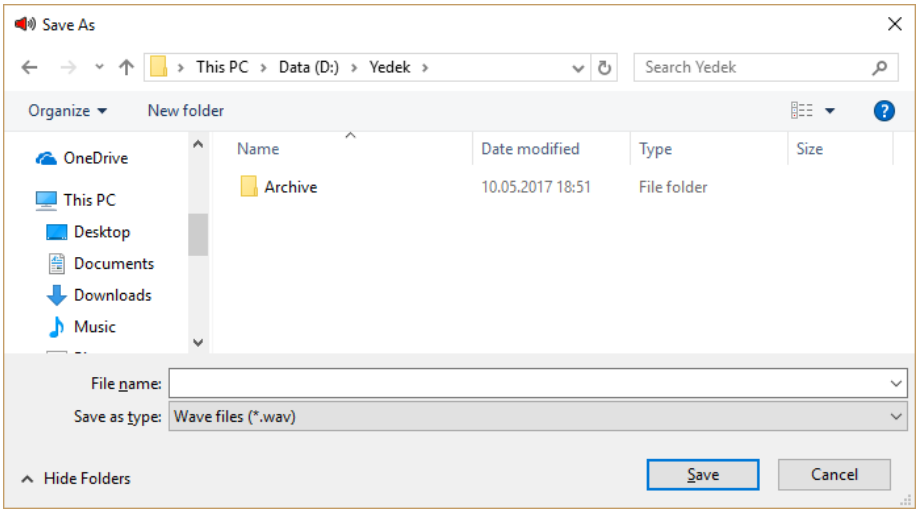


Figure - 17. Save audio file

You can download TekRecorder from KaplanSoft website download section.

Endpoint configuration in TekSIP

You need to create also an endpoint entry for the user's mailbox number in order to redirect calls when users are off-line or busy;

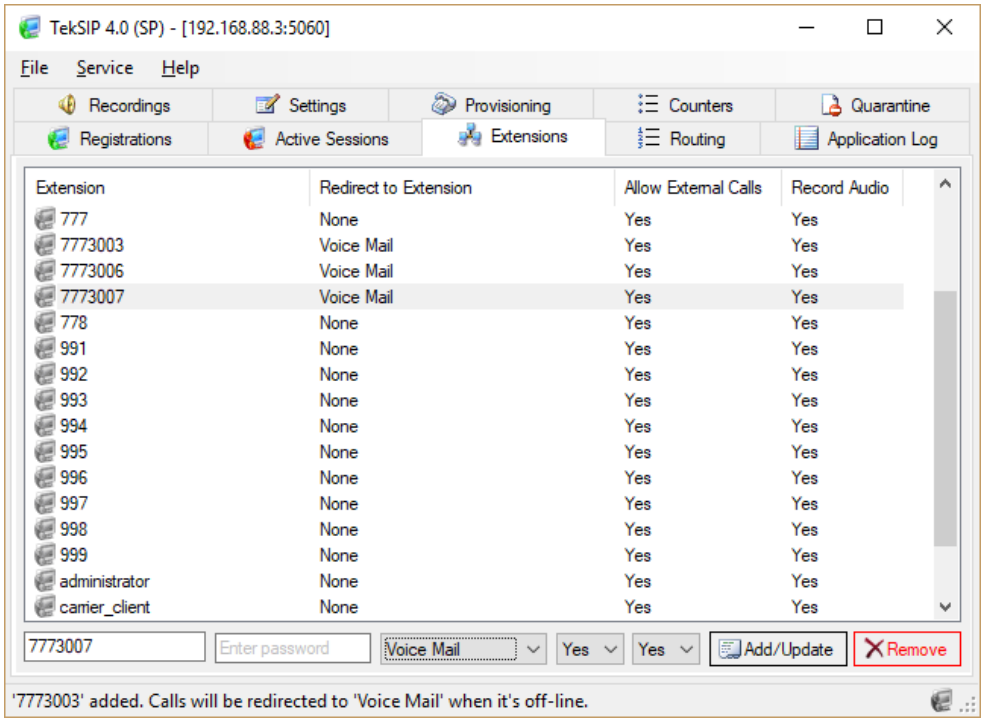


Figure - 18. TekSIP extensions

In the example above extensions 7773006 and 7773007 has mailboxes 7774006 and 7774007 respectively. Both 7774006 and 7774007 are redirected to Voice Mail.

You can see TekIVR endpoint entries for the TekSIP extensions above;

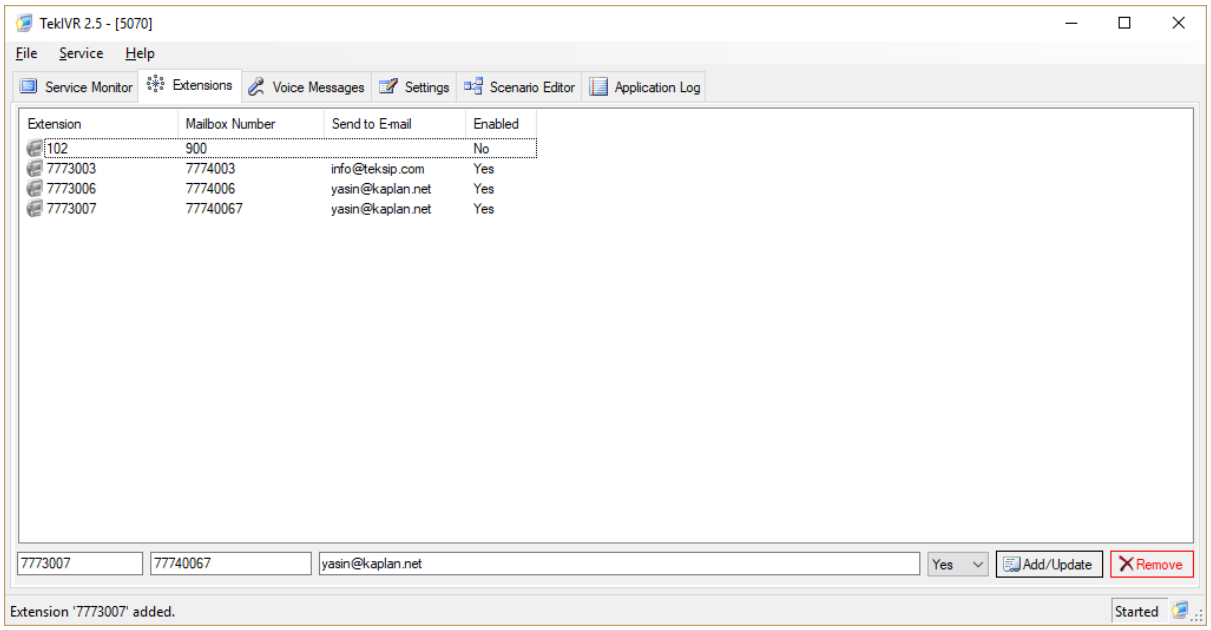


Figure - 19. TekIVR extensions

You need to enter TekIVR IP address and port number in Settings / Services tab to enable Voice Mail re-direction in TekSIP;

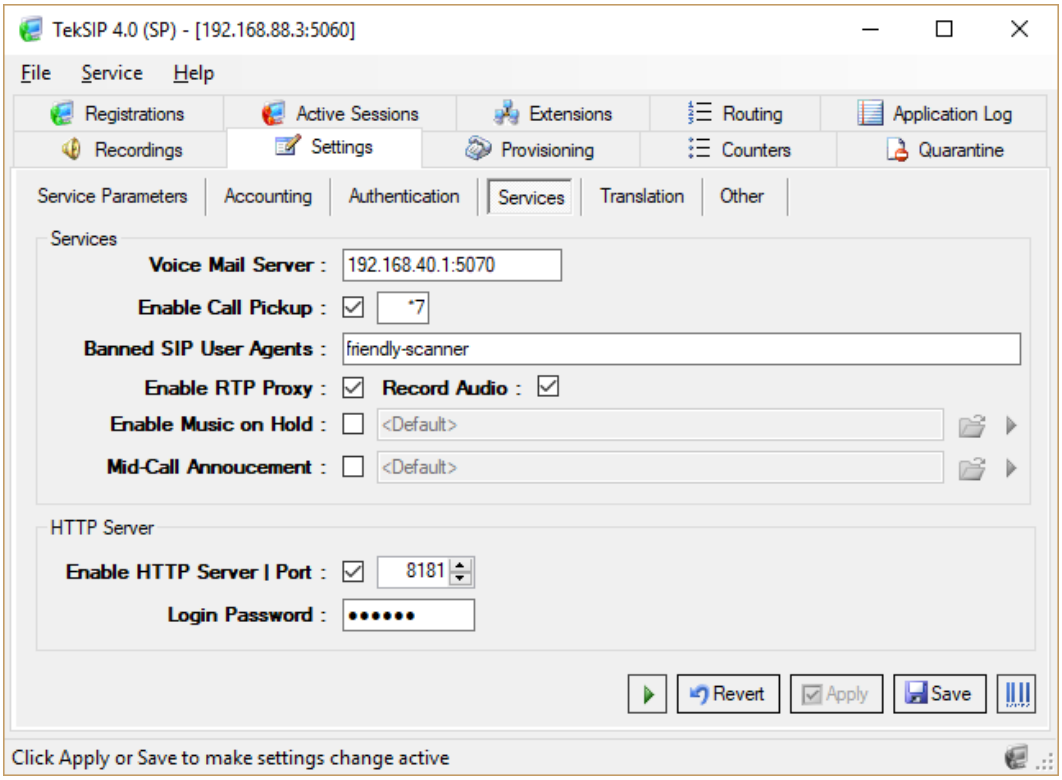
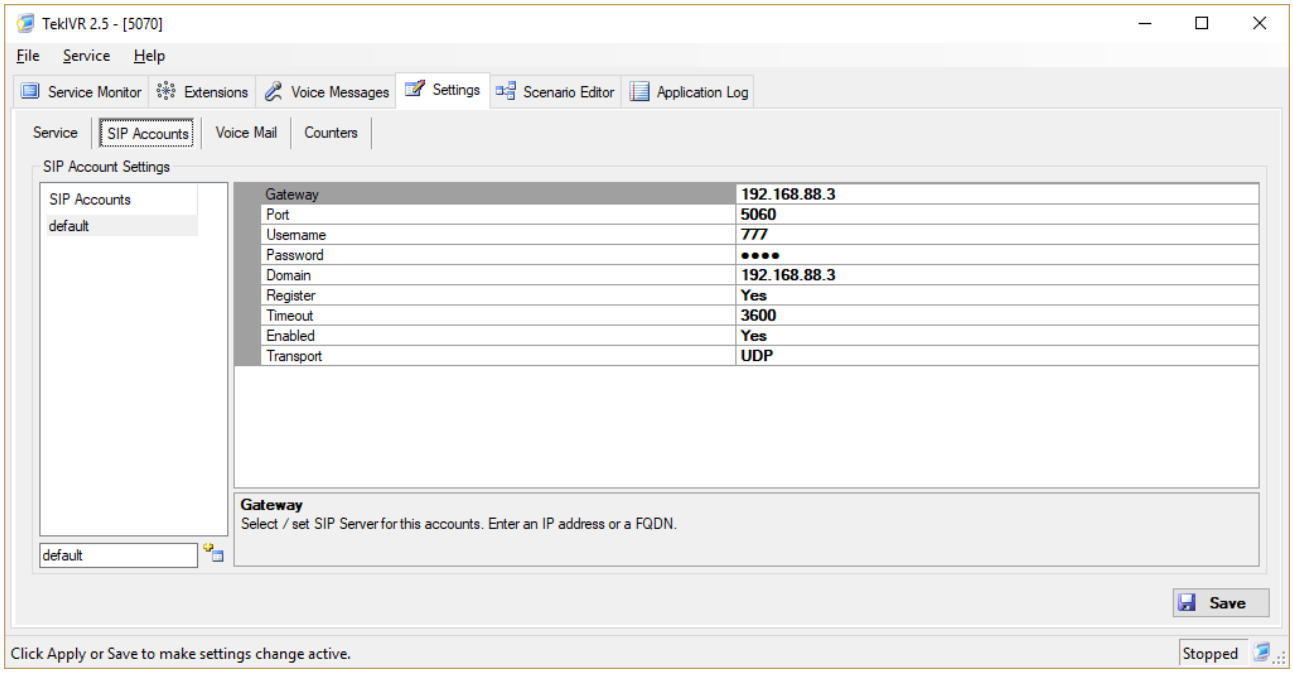


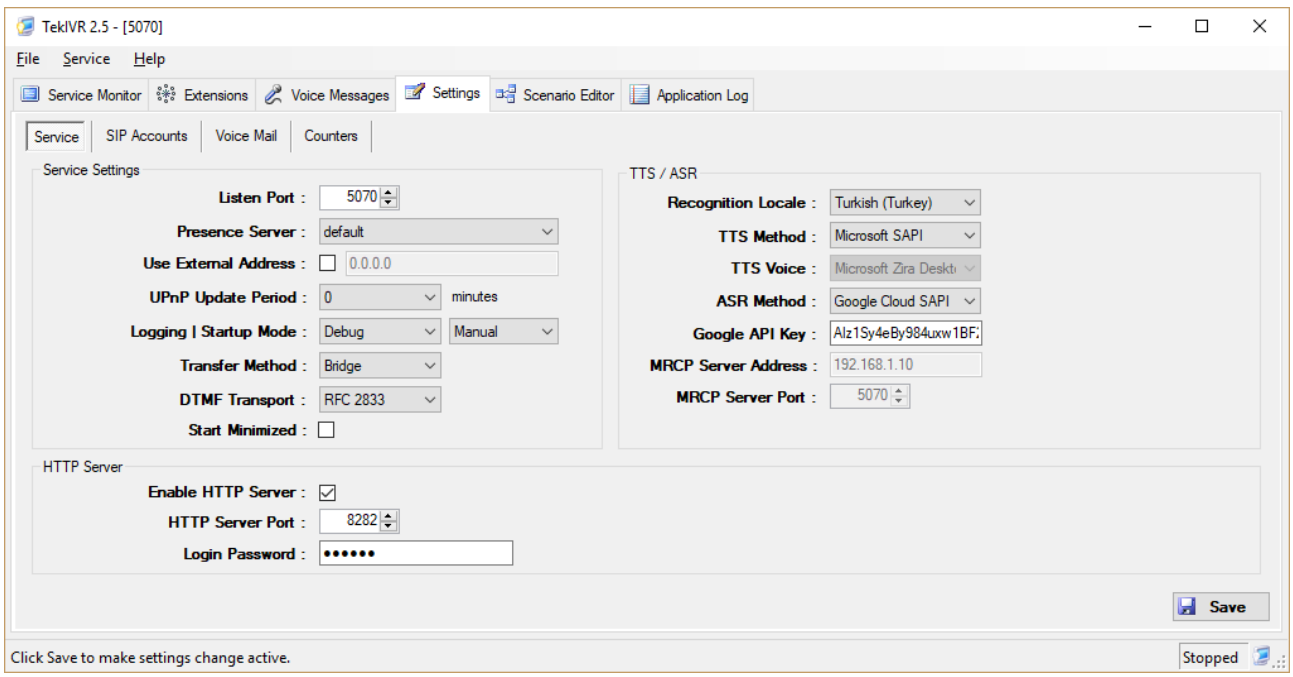
Figure - 20. TekSIP settings

Creating Hunt Groups in Scenarios

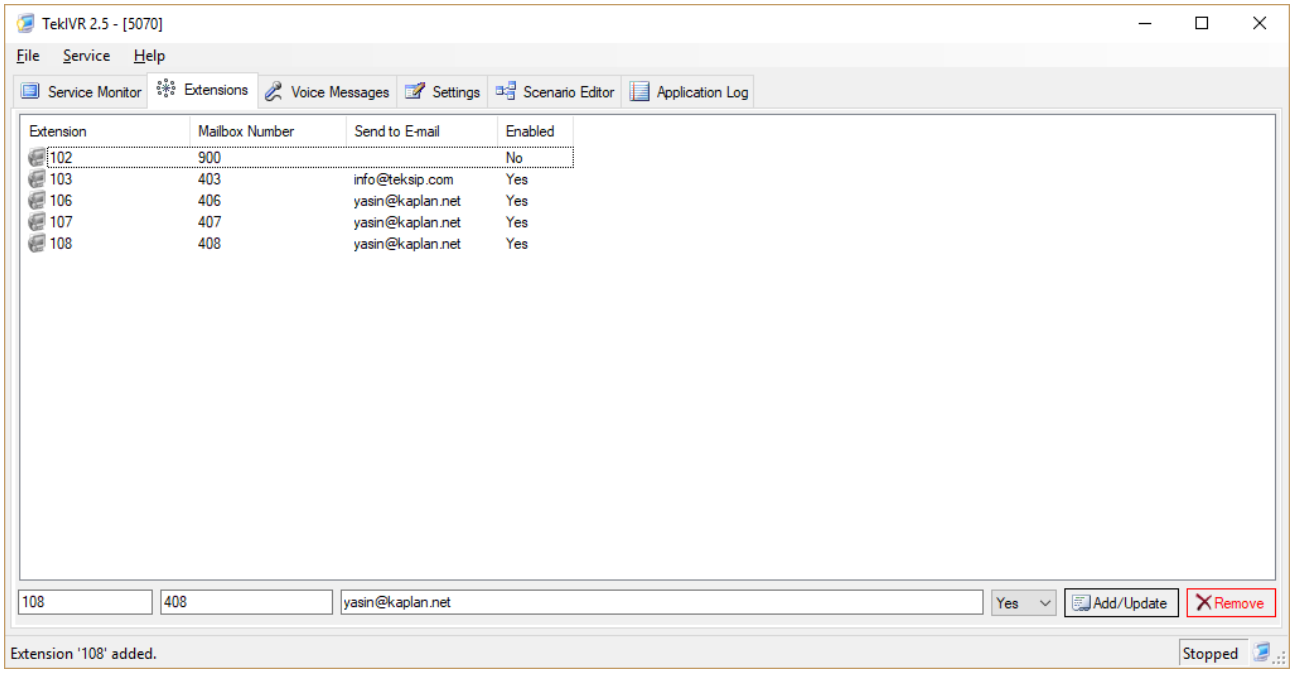
You can create hunt groups to transfer an incoming call to a group of users in TekIVR scenarios. Your SIP server, which all your SIP extensions are registered should act as a SIP Presence Server for this function for a better performance. You should start with creating a SIP account entry for your SIP server.



The next step is specifying this SIP server presence server in TekIVR settings. You must also set the Transfer method to either REFER or Bridge. DTMF method is not supported by hunt groups.

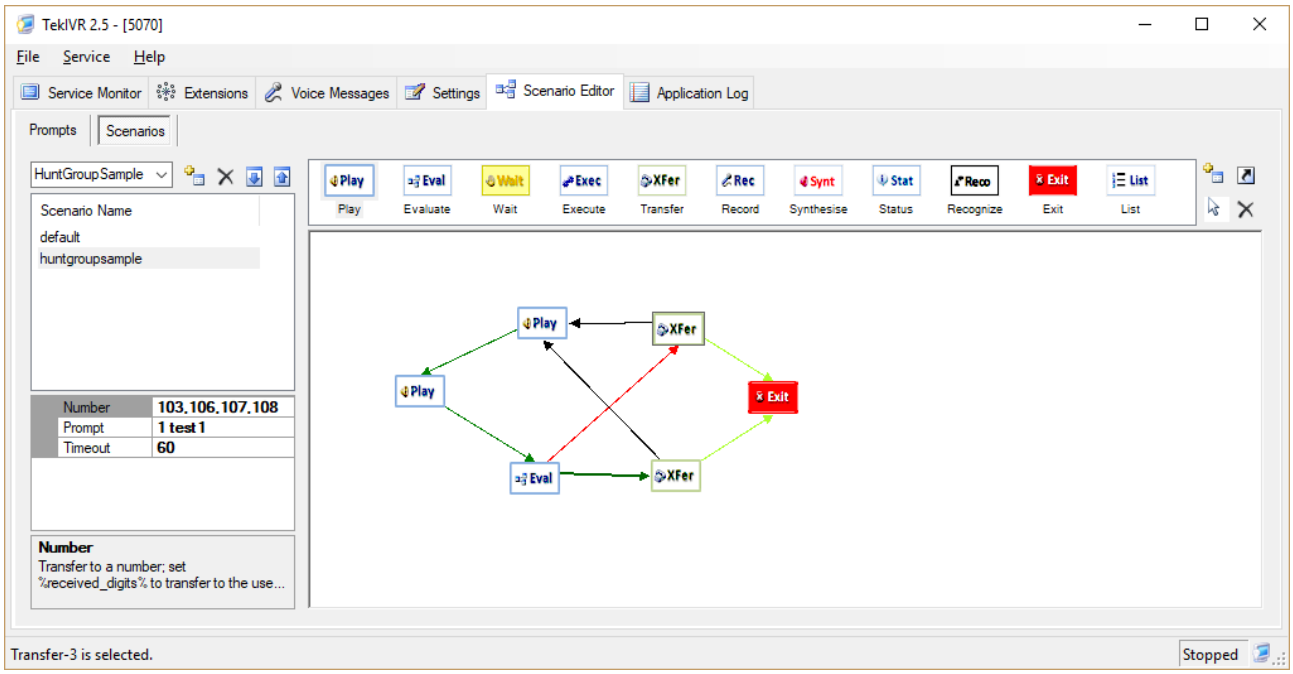


Finally creating extension entries for the endpoints in the hunt group;



Extensions 103, 106, 107, and 108 will form a hunt group in this example. TekIVR will subscribe to the presence status of these extensions. TekIVR will check their presence status and will transfer the incoming call to a randomly selected endpoint from the available endpoints.

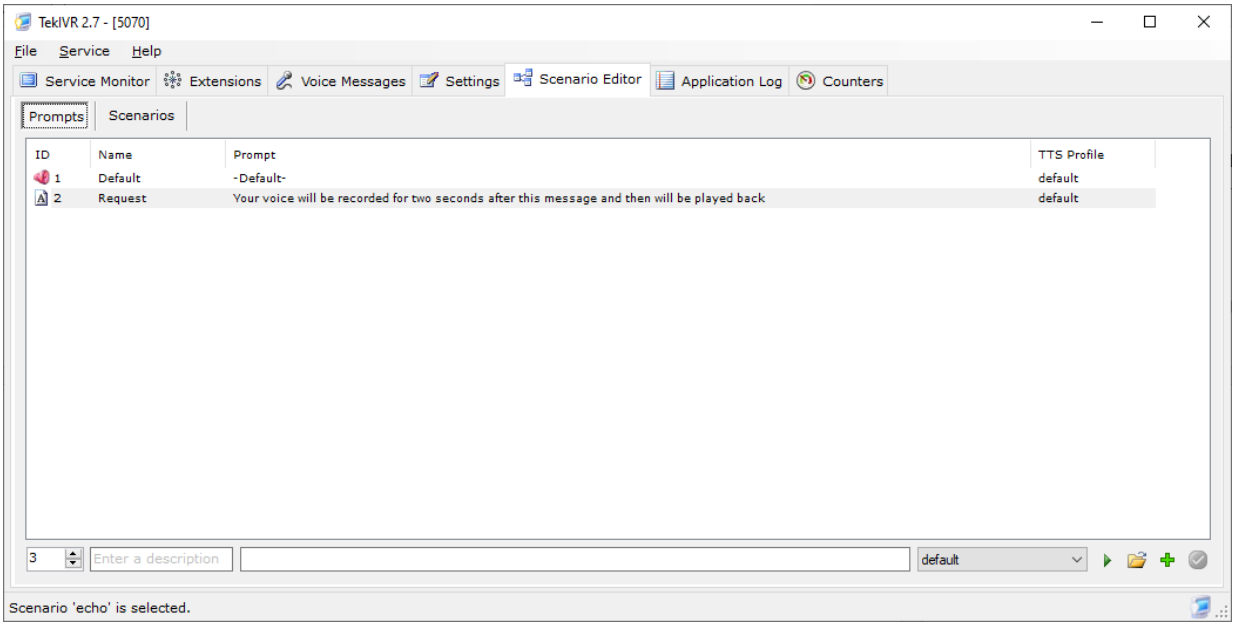
Transfer action in the scenarios will be used for creating hunt groups. Endpoints in the hunt group will be entered to the Number parameter of the Transfer action by concatenating extensions numbers with commas.



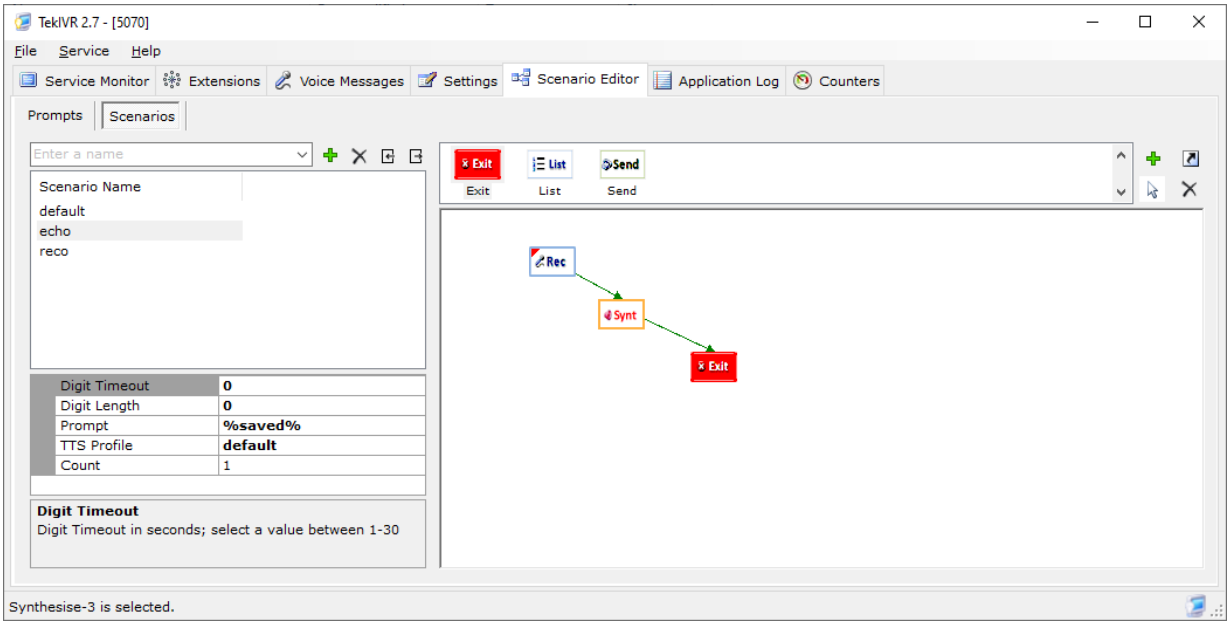
In this scenario the user will be asked for a new destination if the transfer request fails. You can optionally use a cloned transfer action as the next action on a transfer failure, and you can set primary transfer action as the next action on a transfer failure for cloned transfer action.

Echo Test Scenario

You can create a special scenario to test if there is a problem with two-way RTP media flow. Create an informative prompt in Prompts tab first:



Add the following actions to the scenario and link them. Set the prompt above as the prompt for action Rec and set %saved% variable as prompt in action Synt. Recommended Record Duration for action Rec is 2 seconds.



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