mod_httapi

About

This module provides an HTTP based Telephony API using a standard FreeSWITCH application interface as well as a cached http file format interface.

Usage

The file format plugin can be used like this:

```
<action application="playback" data="http://some.com/sounds/foo.wav"/>
```

This syntax is valid anywhere a filename parameter is requested in FreeSWITCH. If your URL contains query parameters (after a "?"), use

```
<action application="record" data="http://(file=/tmp/part1.ul,name=part1.PCMU)example.net/part1.PCMU?rev=47"/>
```

The available parameters are: file, profile, method, name.

The application is called like this:

```
<action application="httapi" data="{url=http://some.host.com/app.cgi}"/>
```

The target url is expected to be a CGI returning text/xml using the documentation below.

The format is roughly as described below (We could use a DTD maybe).

```
<document type="xml/freeswitch-httapi">
 <params>
   <someparam>someval</someparam>
 </params>
 <variables>
   <somevar>someval</somevar>
 </variables>
 <work>
 <...>
 </work>
</document>
```

The profile name must be chosen to bind to a preset series of settings and permissions to use as a basis for the client session. It's chosen from the specified params or from the channel variable using the keyname 'httapi_profile' and if not specified will default to 'default' Any params specified in the initial data of the application encased in curly braces {} will be parsed into the initial parameters similar to originate. These params remain set until the call ends and are supplied as form elements on each hit to the HTTP server. If the permissions allow, a <params> tag is parsed from the resulting document and set into this data set and will be mirrored back on subsequent http posts. Also, if the permissions allow, a <variables> tag is parsed the same way to set channel variables. The <work> tag is required and contains one or more of the supported command tags that generally manipulate the call behavior. The application will continue to do the task in the work section until an error is encountered or when an action from the work tag warrants it. If the session exits the httapi app without hanging up it can do other tasks and make another call to httapi and the session data will remain preserved.

Example

```
<document type="xml/freeswitch-httapi">
 <params></params>
 <work>
  <playback name="exten" file="http://my.com/sounds/exten.wav" error-file="http://my.com/sounds/invalid.wav"
input-timeout="5000">
   <bind strip="#">~\d+#</bind>
  </playback>
 </work>
</document>
```

Note that the value from bind strip is returned to the same page as the parameter exten (taken from name in playback).

Binding

Several of the work tags that indicate they support bindings can contain one or more bind tags that function in similar fashion to bind_digit_action

```
<bind action strip>*EXPR*</bind>
ATTRS:
action              : a specific url to go to next if the binding is dialed
strip               : a character to strip from the result string, such as #
```

NOTE : Srigg reported that multiple <bind strip> tags are needed when you want it to post digits after the timeout expired (in fact, as many tags as the quantity of digits expected)

Work Tag

```
<work>*ACTIONS*</work>

<playback file name error-file action digit-timeout input-timeout loops asr-engine asr-grammar><bind action
strip>*EXPR*</bind></playback>
                    : Plays a file and optionally collects input.
ATTRS:
file                : The file
name                : Parameter name into which to save result
error-file          : Error file to play on invalid input
action              : Change the new target url
temp-action         : Change url to submit to, just for the next loop
digit-timeout       : Pause to wait for digits after file plays (when input bindings are present)
input-timeout       : Pause to wait for more digits in a multi-digit input
loops               : Maximum number of times to play the file when input bindings are present
asr-engine          : Automatic speech recognition engine to use
asr-grammar         : ASR grammar to use

<record file name error-file action digit-timeout input-timeout><bind action strip>*EXPR*</bind></record>
                    : Records a file, optionally collecting input, then posts the file back to the target url.
ATTRS:
file                : The file
name                : Parameter name into which to save result
error-file          : Error file to play on invalid input
action              : Change the new target url
temp-action         : Change url to submit to, just for the next loop
digit-timeout       : Pause to wait for digits after file plays (when input bindings are present)
input-timeout       : Pause to wait for more digits in a multi-digit input

<pause name error-file action digit-timeout input-timeout loops milliseconds><bind action strip>*EXPR*</bind><
/pause>
                    : Waits for input for a specific amount of time.
ATTRS:
milliseconds        : Number of milliseconds to pause
name                : Parameter name into which to save result
error-file          : Error file to play on invalid input
action              : Change the new target url
```

```
temp-action        : Change url to submit to, just for the next loop
digit-timeout      : Pause to wait for digits after file plays (when input bindings are present)
input-timeout      : Pause to wait for more digits in a multi-digit input
loops              : Maximum number of times to play the file when input bindings are present


<speak file name error-file action digit-timeout input-timeout loops engine voice><bind action strip>*EXPR*<
/bind></speak>
                   : Read Text to Speech with optional input.
ATTRS:
file               : The file
name               : Parameter name into which to save result
error-file         : Error file to play on invalid input
action             : Change the new target url
temp-action        : Change url to submit to, just for the next loop
digit-timeout      : Pause to wait for digits after file plays (when input bindings are present)
input-timeout      : Pause to wait for more digits in a multi-digit input
loops              : Maximum number of times to play the file when input bindings are present
engine             : Text-to-speech engine to use
voice              : tts voice to use


<say file name error-file action digit-timeout input-timeout loops language type method gender><bind action
strip>*EXPR*</bind></say>
                   : Use the FreeSWITCH 'say' engine to iterate sounds to simulate a human speaker.
ATTRS:
file               : The file
name               : Parameter name into which to save result
error-file         : Error file to play on invalid input
action             : Change the new target url
temp-action        : Change url to submit to, just for the next loop
digit-timeout      : Pause to wait for digits after file plays (when input bindings are present)
input-timeout      : Pause to wait for more digits in a multi-digit input
loops              : Maximum number of times to play the file when input bindings are present
language           : language to speak
type               : type (fs param)
method             : method (fs param)
gender             : gender (fs param)


<execute application data action>*DATA*</execute>
                   : Execute a FreeSWITCH app.
ATTRS:
application        : The app to run
action             : Change the new target url
temp-action        : Change url to submit to, just for the next loop
data               : Alternate source for app data
*DATA*             : The app data


<sms to action>DATA</sms>
                   : Send a SMS (Short Message Service) message.
ATTRS:
to                 : The destination phone number
action             : Change the new target url
temp-action        : Change url to submit to, just for the next loop
*DATA*             : The message data


<dial context dialplan caller-id-name caller-id-number action>*DATA*</dial>
                   : Place an outbound call or transfer.
ATTRS:
context            : Dialplan context.
dialplan           : Dialplan dialplan.
caller-id-name     : Caller ID Name
caller-id-number   : Caller ID Number
action             : Change the new target url
temp-action        : Change url to submit to, just for the next loop
*DATA*             : Number to dial or originate string


<recordCall limit name action>
                   : Begin recording the call audio.  The file will be posted when the call ends.
ATTRS:
limit              : Timeout in seconds
name               : Name to use for input values
action             : URL action to use
```

```
<conference profile action>
                   : Start a conference call.
ATTRS:
profile            : Conference profile to use
action             : Change the new target url

<hangup cause action>
                   : Hangup the call.
ATTRS:
cause              : Hangup cause
action             : Change the new target url

<break>
                   : Exit the httapi application and continue in the dialplan.

<log level clean action>
                   : Exit the httapi application and continue in the dialplan.
ATTRS:
level              : The log level to use
clean              : If true do not print log prefix
action             : Change the new target url

<continue action>
                   : Just continue (no-op).
ATTRS:
action             : Change the new target url

<getVar action temp-action permanent>
                   : Get a Channel variable (depends on permissions).
ATTRS:
action             : Change the new target url
temp-action        : Change url to submit to, just for the next loop
permanent          : Add as a permanent param or just once

<voicemail action temp-action check auth-only profile domain id>
                   : Check or record voice mail
ATTRS:
action             : Change the new target url
temp-action        : Change url to submit to, just for the next loop
check              : true to check, omit to record a message for this mailbox
auth-only          : authenticate only and continue
profile            : profile name to use, omit for "default"
domain             : domain to use, omit for global domain variable
id                 : mailbox id to use, omit to prompt for id
```

Configuration

```
<settings>:

<param name="" value="">
debug                        : <true|false>         false        Print debug data
file-cache-ttl               : <number of sec>      300          How long to wait before checking the
server to see if audio
file has changed.
file-not-found-expires       : <number of sec>      300          How long to preserve cached audio files
that are not found by the server.

<profile name="<name>">      : CREATE NEW PROFILE TO REFERENCE BY NAME
gateway-url                  : <string>             ""           Initial URL to which to connect
gateway-credentials          : <string>             ""           HTTP credentials
auth-scheme                  : <string              >            basic authentication scheme to use.
[basic|digest|NTLM|GSS-NEGOTIATE|any]
disable-100-continue         : <true|false>         true         Disable the 100 continue feature
method                       : <string>             ""           METHOD name to send
timeout                      : <number>             0            Timeout waiting for response
enable-cacert-check          : <true|false>         false        Check CA/CERT
ssl-cert-path                : <string>             ""           path to file
ssl-cacert-file              : <string>             ""           CA/CERT file
ssl-key-path                 : <string>             ""           path to file
ssl-key-password             : <string>             ""           password to use
ssl-version                  : <string>             ""           ssl version
enable-ssl-verifyhost        : <true|false>         ""           Verify ssl host
cookie-file                  : <string>             ""           Path to file to use for cookie
enable-post-var              : <param_name>         ""           Specify param names ok to send
bind-local                   : <string>             ""           Interface to which to bind
default-profile              : <string>             default      Profile to use when not specified
user-agent                   : <string>             mod_httapi/1.0  User Agent header value

<permissions>: * = default

<permission name="" value="">
*set-params                  : <params> tag can be parsed for session params
set-vars                     : <variables> tag can be parsed to set channel vars
extended-data                : Extended data is sent like full channel event data
execute-apps                 : <execute> tag is enabled to execute apps
expand-vars                  : Allow expansion of FS ${variables} (this opens up all FSAPI calls)
*dial                        : <dial> tag is enabled allowing outbound dialing
dial-set-context             : <dial context=""> context attribute is permitted
dial-set-dialplan            : <dial dialplan=""> dialplan attribute is permitted
dial-set-cid-name            : <dial cid_name=""> cid_name attribute is permitted
dial-set-cid-number          : <dial cid_number=""> cid_number attribute is permitted
dial-full-originate          : <dial> full originate syntax is permitted instead of just a number
*conference                  : <conference> tag is enabled allowing creation of conferences
conference-set-profile       : <conference profile=""> attribute is permitted
all                          : all permissions are set
none                         : no permissions are set
```

Answer and Early Media

When to answer

You don't need to answer the call in the dialplan before calling into httapi. These both extensions below will make httapi requests to my application:

```
    <extension name="test1">
        <condition field="destination_number" expression="^6622$">
            <action application="httapi" data="{url=http://localhost/simver/FSHttApi/Test}" />
        </condition>
    </extension>

    <extension name="test2">
        <condition field="destination_number" expression="^6621$">
            <action application="answer" />
            <action application="httapi" data="{url=http://localhost/simver/FSHttApi/Test}" />
        </condition>
    </extension>
```

You can play back sounds or 'say' something without explicitly 'answering' the call. But the call is early media only and in this state you won't be able to collect dtmf, for example. If you want to interact with the call you have to answer then since early media is an intermediate state and doesn't mean the call will stay alive for much long.

Answering calls

You can answer the call with <application name="answer"/> but by default there's no permission to run the 'answer' application from httapi script so you need to enable it in httapi.conf.xml.