

schmupu / ioBroker.asterisk Public

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master



ioBroker.asterisk / README.md



schmupu Issue #8 fix



3 contributors



ioBroker Asterisk VoIP Adapter

build error build failing installed 617 stable v1.0.6 npm v1.0.6 downloads 90/month

```
npm install iobroker.asterisk
```

[German manual / Deutsche Anleitung](#)

The Asterisk adapter converts text messages to audio files and calls then over Asterisk by VoIP any telephone number you want and plays the audio message.

Install / Configurations

Asterisk has to connect for outgoing calls with your voip provider like Telekom or Vodafone or with your FritzBox! Please follow one of these installation guides.

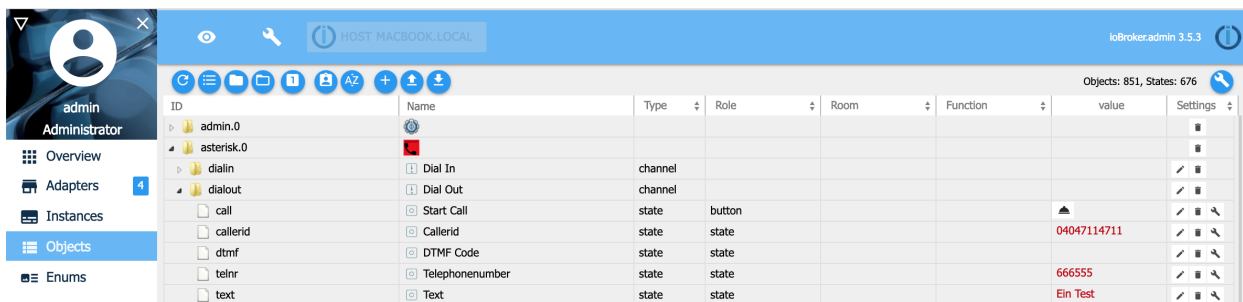
- Configuration [Asterisk via SIP with the FritzBox](#) (the easiest way)
- Configuration [Asterisk via PJSIP with the FritzBox](#) (pjsip is more modern as sip)
- Configuration [Asterisk via PJSIP with Telekom as provider](#)
- Configuration [Asterisk via PJSIP with Sipgate as provider](#)
- Configuration [ssh/scp](#) (ioBroker and asterisk runs on different server)

Using Asterisk

Using Asterisk with objects / states for dialing out

The easiest way to use asterisk is through the ioBroker objects page. There, fill the following values under dialout parameter:

- call: push button to initiate a call
- callerid: telephonenumber which will be shown the callee
- dtmf: the callee pressed numbers on the keypad
- telnr: the number to be dialed
- text: the text that will be played to the callee

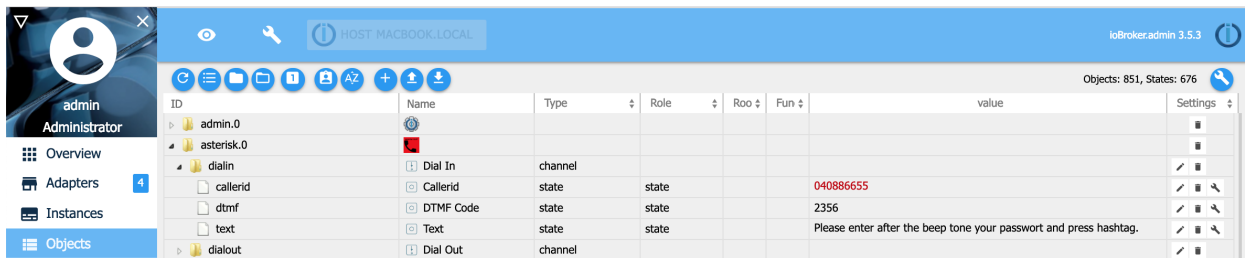


ID	Name	Type	Role	Room	Function	value	Settings
admin.0							
asterisk.0							
dialin	Dial In	channel					
dialout	Dial Out	channel					
call	Start Call	state	button				
callerid	Callerid	state	state			04047114711	
dtmf	DTMF Code	state	state				
telnr	Telephonenumber	state	state			666555	
text	Text	state	state			Ein Test	

Using Asterisk with objects / states for dialing in

If you configured your SIP Provider (for example Fritzbox, Sipgate, ...) and the Asterisk Configuration to allow dialin calls you can set following parameter

- callerid: telephonenumber which called asteriks
- dtmf: callers pressed numbers on the keypad
- text: the text that will be played to the caller



ID	Name	Type	Role	Ro	Fun	value	Settings
admin.0							
asterisk.0							
dialin	Dial In	channel					
callerid	Callerid	state	state			040886655	
dtmf	DTMF Code	state	state			2356	
text	Text	state	state			Please enter after the beep tone your password and press hashtag.	
dialout	Dial Out	channel					

Using Asterisk with javascript or blocky for dialing out

Now you can use the adapter in your javascript or blocky programm.

```

var number = "040 666-7766";
var callerid = '040 123 999'; // optional, if not set anonymous call
var msg = "Hello, this textmessage will be converted to audio";

// call telephone number 040 666-7766 and play text message as audio
sendTo('asterisk.0', "dial", { telnr: number, callerid: callerid, text: msg}, (
  console.log('Result: ' + JSON.stringify(res));
});

// call telephone number 040 666-7766 and play mp3 audio file
// mp3 file has to exist on asterix server
sendTo('asterisk.0', "dial", { telnr: number, callerid: callerid, aufiofile: '/tr
  console.log('Result: ' + JSON.stringify(res));
});

// call telephone number 040 666-7766 and play gsm audio file
// gsm file has to exist on asterix server
sendTo('asterisk.0', "dial", { telnr: number, callerid: callerid, aufiofile: '/tr
  console.log('Result: ' + JSON.stringify(res));
});

// Show entered DTMF code
on({ id: "asterisk.0.dialin.dtmf"/*DTMF Code*/ }, (obj) => {
  let dtmf = obj.state.val;
  console.log("DTMF: " + dtmf);
});

// Show entered DTMF code
on({ id: "asterisk.0.dialout.dtmf"/*DTMF Code*/ }, (obj) => {
  let dtmf = obj.state.val;
  console.log("DTMF: " + dtmf);
});

```

You can use following parameter in the sendTo dial statement:

- **language:** language take for text to speech (tts) function. (allowed values: 'DE', 'EN', ... Default is the ioBroker system language)

- **repeat:** how many times shall the audio message repeated (allowed values 1 to n, default 5)
- **priority:** if you send parallel many sendTo dial statements, the messages with a smallest priority will be send first (allowed values 1 to n, default 1)
- **text:** text message that will be send as audio

☰ Executable File | 130 lines (94 sloc) | 6.55 KB



- **async:** Allows multiple calls to be generated without waiting for a response (allowed values: false/true, default false)
- **audiofile:** if you using the text parameter. The converted text to audio will be saved in audiofile. If the audiofile exist, it will be overwritten. If you do not use the parameter text, the audiofile will be played.
- **callerid:** Defines the identifier (your sender telephone number) . If callerid is missing the transferred telephone number will be anonymous

Resolving problems

If you have problems with asterisk, you can try to find something in the logfiles under /var/log/asterisk. After you started asterisk you can call asterisk with asterisk -rvvvvv on the comand shell for debugging. After you started asterisk -rvvvvv you can initialize a call by iobroker and see what happens.

🔗 Changelog

[Changelog](#)

License

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