
pytgvoip Documentation

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Telegram VoIP Library for Python

Community

PytgVoIP is a [Telegram](#) VoIP library written in Python and C++.

It uses [libtgvoip](#) (a library used in official clients) for voice encoding and transmission, and [pybind11](#) for simple generation of Python extension written in C++.

CHAPTER 1

Features

- Making and receiving Telegram calls
- Python callbacks for sending and receiving audio stream frames allow flexible control
- Pre-built Windows wheels in PyPI

CHAPTER 2

Requirements

- Python 3.4 or higher
- An MTPROTO client (i.e. Pyrogram, Telethon)

CHAPTER 3

Installing

Refer the corresponding section: *Installation*

CHAPTER 4

Encoding audio streams

Streams consumed by `libtgvoip` should be encoded in 16-bit signed PCM audio.

```
$ ffmpeg -i input.mp3 -f s16le -ac 1 -ar 48000 -acodec pcm_s16le input.raw # encode
$ ffmpeg -f s16le -ac 1 -ar 48000 -acodec pcm_s16le -i output.raw output.mp3 # decode
```


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5.1 Installation

5.1.1 Requirements

On Linux and macOS to install this library you must have `cmake`, C++11 compatible compiler, Python headers and `libtgvoip` installed:

- Debian-based distributions

```
$ apt install cmake gcc g++ python3-dev
```

- Archlinux-based distributions

```
$ pacman -S cmake gcc python3
```

- macOS

```
$ brew install cmake gcc g++ python3
```

For instructions on building `libtgvoip` refer to the corresponding docs section: [Building libtgvoip](#)

5.1.2 Install pytgvoip

- Stable version:

```
$ pip install pytgvoip
```

- Development version:

```
$ pip install git+https://github.com/bakatrouble/pytgvoip#egg=pytgvoip
```

5.2 Building libtgvoip

5.2.1 Requirements

- Debian-based distributions

```
$ apt install make autoconf automake gcc g++ openssl libssl-dev libopus0 libopus-  
↳dev
```

- Archlinux-based distributions

```
$ pacman -S make autoconf automake gcc openssl opus
```

- macOS

```
$ brew install make autoconf automake gcc g++ openssl opus
```

5.2.2 Build and install

```
$ cd /tmp  
$ git clone https://github.com/grishka/libtgvoip/  
$ cd libtgvoip  
$ git checkout b6ac2911 # confirmed to work with this version, others would require_  
↳testing  
  
$ export CFLAGS="-O3"  
$ export CXXFLAGS="-O3"  
$ autoreconf --force --install  
$ ./configure --enable-audio-callback --enable-static=no  
$ make # add "-jN" flag for multithreaded build, N=(cpu core count + 1) is_  
↳recommended  
$ make install
```

5.3 Usage

5.3.1 Common

- You should have a protocol object: `phoneCallProtocol(min_layer=65, max_layer=VoIPController.CONNECTION_MAX_LAYER, udp_p2p=True, udp_reflector=True)`
- All VoIP-related updates have type of `updatePhoneCall` with `phone_call` field of types `phoneCallEmpty`, `phoneCallWaiting`, `phoneCallRequested`, `phoneCallAccepted`, `phoneCall` or `phoneCallDiscarded`
- Use `tgvoip.utils.generate_visualization()` with `auth_key` and `g_a` for outgoing or `g_a_or_b` for incoming calls to get emojis if you need them

5.3.2 Starting conversation

- Create a `VoIPController` instance
- Call `tgvoip.VoIPController.set_send_audio_frame_callback()` (see docs for arguments) if needed, otherwise silence will be sent
- Call `tgvoip.VoIPController.set_recv_audio_frame_callback()` (see docs for arguments) if needed, otherwise nothing will be done to incoming audio stream
- Add state change handlers to `tgvoip.VoIPController.call_state_changed_handlers` (see docs for handler format) list if needed
- Add signal bars change handlers to `tgvoip.VoIPController.signal_bars_changed_handlers` (see docs for handler format) list if needed
- Invoke `help.getConfig()` (result is later referred as `config`)
- Call `tgvoip.VoIPController.set_config()` (arguments are: `config.call_packet_timeout_ms / 1000.`, `config.call_connect_timeout_ms / 1000.`, `DataSaving.NEVER`, `call.id`)
- Call `tgvoip.VoIPController.set_encryption_key()` (arguments are: `i2b(auth_key)`, `is_outgoing` where `is_outgoing` is a corresponding boolean value)
- Build a list of `tgvoip.Endpoint` objects from `call.connection` (single) and `call.alternative_connections` (another list)
- Call `tgvoip.VoIPController.set_remote_endpoints()` (arguments are: `endpoints`, `call.p2p_allowed`, `False`, `call.protocol.max_layer`)
- Call `tgvoip.VoIPController.start()`
- Call `tgvoip.VoIPController.connect()`

5.3.3 Discarding call

- Build peer: `inputPhoneCall(id=call.id, access_hash=call.access_hash)`
- Get call duration using `tgvoip.VoIPController.call_duration`
- Get connection ID using `tgvoip.VoIPController.get_preferred_relay_id()`
- Build a suitable reason object (types are: `phoneCallDiscardReasonBusy`, `phoneCallDiscardReasonDisconnect`, `phoneCallDiscardReasonHangup`, `phoneCallDiscardReasonMissed`)
- Invoke `phone.discardCall(peer, duration, connection_id, reason)`. You might get `CALL_ALREADY_DECLINED` error, this is fine
- Destroy the `tgvoip.VoIPController` object

5.3.4 Ending conversation

- Send call rating and debug log if call ended normally (not failed): *TBD*
- Destroy the `tgvoip.VoIPController` object, everything will be done automatically

5.3.5 Making outgoing calls

- Get a `user_id` object for user you want to call (of type `inputPeerUser`)
- Request a Diffie-Hellman config using `messages.getDhConfig(version=0, random_length=256)`
- Check received config using `tgvoip.utils.check_dhc()`. If check is not passed, do not make the call. You might want to cache received config because check is expensive
- Choose a random value `a`, $1 < a < dhc.p-1$
- Calculate `g_a`: `pow(dhc.g, a, dhc.p)`
- Calculate `g_a_hash`: `sha256(g_a)`
- Choose a random value `random_id`, $0 \leq random_id \leq 0x7fffffff-1$
- Invoke `phone.requestCall(user_id, random_id, g_a_hash, protocol)`
- Wait for an update with `phoneCallAccepted` object, it means that other party has accepted the call. You also might get a `phoneCallDiscarded` object, it means that other party has declined the call
- If you have got a `phoneCallDiscarded` object, stop the `tgvoip.VoIPController`. Otherwise, continue
- Check a `g_b` value from received `phoneCallAccepted` (later referred as `call`) object using `tgvoip.utils.check_g()`. If check is not passed, stop the call
- Calculate `auth_key`: `pow(call.g_b, a, dhc.p)`
- Calculate `key_fingerprint` using `tgvoip.utils.calc_fingerprint()`
- Build `peer`: `inputPhoneCall(id=call.id, access_hash=call.access_hash)`
- Invoke `phone.confirmCall(key_fingerprint, peer, g_a, protocol)`
- Start the conversation

5.3.6 Receiving calls

- You will receive an update containing `phoneCallRequested` object (later referred as `call`). You might discard it right away (use 0 for duration and `connection_id`)
- Request a Diffie-Hellman config using `messages.getDhConfig(version=0, random_length=256)`
- Check received config using `tgvoip.utils.check_dhc()`. If check is not passed, do not make the call. You might want to cache received config because check is expensive
- Choose a random value `b`, $1 < b < dhc.p-1$
- Calculate `g_b`: `pow(dhc.g, b, dhc.p)`
- Save `call.g_a_hash`
- Build `peer`: `inputPhoneCall(id=call.id, access_hash=call.access_hash)`
- Invoke `phone.acceptCall(peer, g_b, protocol)`. You might get `CALL_ALREADY_DISCARDED` or `CALL_ALREADY_ACCEPTED` errors, then you should stop current conversation. Also, if response contains `phoneCallDiscarded` object you should stop the call
- Wait for an update with `phoneCall` object (later referred as `call`)

- Check that `call.g_a_or_b` is not empty and `sha256(call.g_a_or_b)` equals to `g_a_hash` you saved before. If it doesn't match, stop the call
- Check a `call.g_a_or_b` value object using `tgvoip.utils.check_g()` (second argument is `dhc.p`). If check is not passed, stop the call
- Calculate `auth_key`: `pow(call.g_a_or_b, b, dhc.p)`
- Calculate `key_fingerprint` using `tgvoip.utils.calc_fingerprint()`
- Check that `key_fingerprint` you have just calculated matches `call.key_fingerprint`. If it doesn't match, stop the call
- Start the conversation

5.4 libtgvoip wrapper

5.4.1 VoIPController

class `tgvoip.VoIPController` (*persistent_state_file*: str = "", *debug*=False, *logs_dir*='logs')

A wrapper around C++ wrapper for libtgvoip VoIPController

Parameters

- **persistent_state_file** (str, optional) – ?, empty to not use
- **debug** (bool, optional) – Modifies logging behavior
- **logs_dir** (str, optional) – Logs directory

Class attributes:

LIBTGVOIP_VERSION Used libtgvoip version

CONNECTION_MAX_LAYER Maximum layer supported by used libtgvoip version

persistent_state_file

Value set in the constructor

call_state_changed_handlers

list of call state change callbacks, callbacks receive a `CallState` object as argument

signal_bars_changed_handlers

list of signal bars count change callbacks, callbacks receive an `int` object as argument

call_duration

Current call duration in seconds as `int` if call was started, otherwise 0

start ()

Start the controller

connect ()

Start the call

set_proxy (*address*: str, *port*: int = 1080, *username*: str = "", *password*: str = "")

Set SOCKS5 proxy config

Parameters

- **address** (str) – Proxy hostname or IP address
- **port** (int, optional) – Proxy port

- **username** (int, *optional*) – Proxy username
- **password** (int, *optional*) – Proxy password

Raises ValueError if address is empty

set_encryption_key (*key: bytes, is_outgoing: bool*)

Set call auth key

Parameters

- **key** (bytes) – Auth key, must be exactly 256 bytes
- **is_outgoing** (bool) – Is call outgoing

Raises ValueError if provided auth key has wrong length

set_remote_endpoints (*endpoints: List[<sphinx.ext.autodoc.importer.MockObject object at 0x7f7b304f3588>], allow_p2p: bool, tcp: bool, connection_max_layer: int*)

Set remote endpoints received in call object from Telegram.

Usually it's [call.connection] + call.alternative_connections.

You must build Endpoint objects from MTPProto phoneConnection objects and pass them in list.

Parameters

- **endpoints** (list of Endpoint) – List of endpoints
- **allow_p2p** (bool) – Is p2p connection allowed, usually call.p2p_allowed value is used
- **tcp** (bool) – Connect via TCP, not recommended
- **connection_max_layer** (int) – Use a value provided by VoIPController.CONNECTION_MAX_LAYER

Raises ValueError if either no endpoints are provided or endpoints without IPv4 or with wrong peer_tag (must be either None or have length of 16 bytes) are detected

get_debug_string () → str

Get debug string

Returns: str containing debug info

set_network_type (*_type: tgvoip.tgvoip.NetType*)

Set network type

Parameters **_type** (NetType) – Network type to set

set_mic_mute (*mute: bool*)

Set “microphone” state. If muted, audio is not being sent

Parameters **mute** (bool) – Whether to mute “microphone”

set_config (*recv_timeout: float, init_timeout: float, data_saving_mode: tgvoip.tgvoip.DataSaving, call_id: int, enable_aec: bool = True, enable_ns: bool = True, enable_agc: bool = True, log_file_path: str = None, status_dump_path: str = None, log_packet_stats: bool = None*)

Set call config

Parameters

- **recv_timeout** (float) – Packet receive timeout, usually value received from help.getConfig() is used

- **init_timeout** (float) – Packet init timeout, usually value received from `help.getConfig()` is used
- **data_saving_mode** (*DataSaving*) – Data saving mode
- **call_id** (int) – Call ID
- **enable_aec** (bool, *optional*) – Whether to enable automatic echo cancellation, defaults to True
- **enable_ns** (bool, *optional*) – Whether to enable noise suppression, defaults to True
- **enable_agc** (bool, *optional*) – Whether to enable automatic gain control, defaults to True
- **log_file_path** (str, *optional*) – Call log file path, calculated automatically if not provided
- **status_dump_path** (str, *optional*) – Status dump path, calculated automatically if not provided and debug is enabled
- **log_packet_stats** (bool, *optional*) – Whether to log packet stats, defaults to debug value

debug_ctl (*request: int, param: int*)
Debugging options

Parameters

- **request** (int) – Option (1 for max bitrate, 2 for packet loss (in percents), 3 for toggling p2p, 4 for toggling echo cancelling)
- **param** (int) – Numeric value for options 1 and 2, 0 or 1 for options 3 and 4

get_preferred_relay_id() → int
Get preferred relay ID (used in `discardCall` MTPProto request)

Returns int ID

get_last_error() → `tgvoip.tgvoip.CallError`
Get last error type

Returns *CallError* matching last occurred error type

get_stats() → `<sphinx.ext.autodoc.importer._MockObject object at 0x7f7b304f3630>`
Get call stats

Returns Stats object

get_debug_log() → str
Get debug log

Returns JSON str containing debug log

set_audio_output_gain_control_enabled (*enabled: bool*)
Toggle output gain control

Parameters **enabled** (bool) – Whether to enable output gain control

set_echo_cancellation_strength (*strength: int*)
Set echo cancellation strength, does nothing currently but was in Java bindings (?)

Parameters **strength** (int) – Strength value

get_peer_capabilities() → int
Get peer capabilities

Returns `int` with bit mask, looks like it is used only for experimental features (group, video calls)

need_rate () → `bool`

Get whether the call needs rating

Returns `bool` value

update_state (*state: tgvoip.tgvoip.CallState*)

Manually update state (only triggers handlers)

Parameters *state* (*CallState*) – State to set

set_send_audio_frame_callback (*func: callable*)

Set callback providing audio data to send

Should accept one argument (`int` length of requested audio frame) and return `bytes` object with audio data encoded in 16-bit signed PCM

If returned object has insufficient length, it will be automatically padded with zero bytes

Parameters *func* (*callable*) – Callback function

set_recv_audio_frame_callback (*func: callable*)

Set callback receiving incoming audio data

Should accept one argument (`bytes`) with audio data encoded in 16-bit signed PCM

Parameters *func* (*callable*) – Callback function

5.4.2 VoIPServerConfig

class `tgvoip.VoIPServerConfig` (**args, **kwargs*)

Global server config class. This class contains default config in its source

classmethod `set_config` (*_json: Union[str, dict]*)

Set global server config

Parameters *_json* (*str* | *dict*) – either JSON-encoded object or `dict` containing config values. Might be received from `MTPProto.phone.getCallConfig()` call, if not set default values are used

Raises Prints an error to `stderr` if JSON parsing (for `str` argument) or encoding (for `dict` argument) has occurred

classmethod `set_bitrate_config` (*init_bitrate: int = 16000, max_bitrate: int = 20000, min_bitrate: int = 8000, decrease_step: int = 1000, increase_step: int = 1000*)

Helper method for setting bitrate options

Parameters

- **init_bitrate** (*int*) – Initial bitrate value
- **max_bitrate** (*int*) – Maximum bitrate value
- **min_bitrate** (*int*) – Minimum bitrate value
- **decrease_step** (*int*) – Bitrate decrease step
- **increase_step** (*int*) – Bitrate increase step

Raises Same as `set_config()`

5.4.3 Enums

class `tgvoip.NetType`

An enumeration of network types

Members:

- UNKNOWN = 0
- GPRS = 1
- EDGE = 2
- NET_3G = 3
- HSPA = 4
- LTE = 5
- WIFI = 6
- ETHERNET = 7
- OTHER_HIGH_SPEED = 8
- OTHER_LOW_SPEED = 9
- DIALUP = 10
- OTHER_MOBILE = 11

class `tgvoip.DataSaving`

An enumeration of data saving modes

Members:

- NEVER = 0
- MOBILE = 1
- ALWAYS = 2

class `tgvoip.CallState`

An enumeration of call states

Members:

- WAIT_INIT = 1
- WAIT_INIT_ACK = 2
- ESTABLISHED = 3
- FAILED = 4
- RECONNECTING = 5
- HANGING_UP = 10
- ENDED = 11
- EXCHANGING_KEYS = 12
- WAITING = 13
- REQUESTING = 14
- WAITING_INCOMING = 15
- RINGING = 16

- `BUSY = 17`

class `tgvoip.CallError`
An enumeration of call errors

Members:

- `UNKNOWN = 0`
- `INCOMPATIBLE = 1`
- `TIMEOUT = 2`
- `AUDIO_IO = 3`
- `PROXY = 4`

5.4.4 Data structures

class `tgvoip.Stats`
Object storing call stats

bytes_sent_wifi
Amount of data sent over WiFi :type: int

bytes_sent_mobile
Amount of data sent over mobile network :type: int

bytes_rcvd_wifi
Amount of data received over WiFi :type: int

bytes_rcvd_mobile
Amount of data received over mobile network :type: int

class `tgvoip.Endpoint`
Object storing endpoint info

Parameters

- **`_id`** (`int`) – Endpoint ID
- **`ip`** (`str`) – Endpoint IPv4 address
- **`ipv6`** (`str`) – Endpoint IPv6 address
- **`port`** (`int`) – Endpoint port
- **`peer_tag`** (`bytes`) – Endpoint peer tag

5.5 Utility functions

`tgvoip.utils.i2b` (*value: int*) → bytes
Convert integer value to bytes

Parameters **value** (`int`) – Value to convert

Returns Resulting `bytes` object

`tgvoip.utils.b2i` (*value: bytes*) → int
Convert bytes value to integer

Parameters **value** (`bytes`) – Value to convert

Returns Resulting `int` object

`tgvoip.utils.check_dhc(g: int, p: int) → None`

Security checks for Diffie-Hellman prime and generator. Ported from Java implementation for Android

Parameters

- **g** (`int`) – DH generator
- **p** (`int`) – DH prime

Raises `ValueError` if checks are not passed

`tgvoip.utils.check_g(g_x: int, p: int) → None`

Check `g_numbers`

Parameters

- **g_x** – `g_number` to check
- **p** – DH prime

Raises `ValueError` if checks are not passed

`tgvoip.utils.calc_fingerprint(key: bytes) → int`

Calculate key fingerprint

Parameters **key** (`bytes`) – Key to generate fingerprint for

Returns `int` object representing a key fingerprint

`tgvoip.utils.generate_visualization(key: Union[bytes, int], part2: Union[bytes, int]) -> (typing.List[str], typing.List[str])`

Generate emoji visualization of key

<https://core.telegram.org/api/end-to-end/voice-calls#key-verification>

Parameters

- **key** (`bytes` | `int`) – Call auth key
- **part2** (`bytes` | `int`) – `g_a` value of the caller

Returns A tuple containing two lists (of emoji strings and of their text representations)

`tgvoip.utils.get_real_elapsed_time() → float`

Get current performance counter value

Returns Time to use for measuring call duration

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