

Installation guide

In order to use Video Chat (Web Conferencing) you need to setup STUN Servers. You can use free public solutions or setup your own private server using existing servers in your company.

Any more, you can register in any STUN/TURN cloud providers (like [twilio.com](https://www.twilio.com) or [xirsys.com](https://www.xirsys.com)) and use their STUN server to setup your own torrent server.

Using your recommendations of specific service that you use, get your "iceServers" JSON file, copy it and paste to Video Chat (Video_Chat__c) custom object record in *RTC Connection Settings* field. Note: keep only one Video Chat record in this object. The format of JSON file you can find in Documentation part of this file.

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Buttons: New Import Search this list...

New Video Chat

Information

Video Chat Number	Owner
	Sergey Kozak

RTC Connection Settings

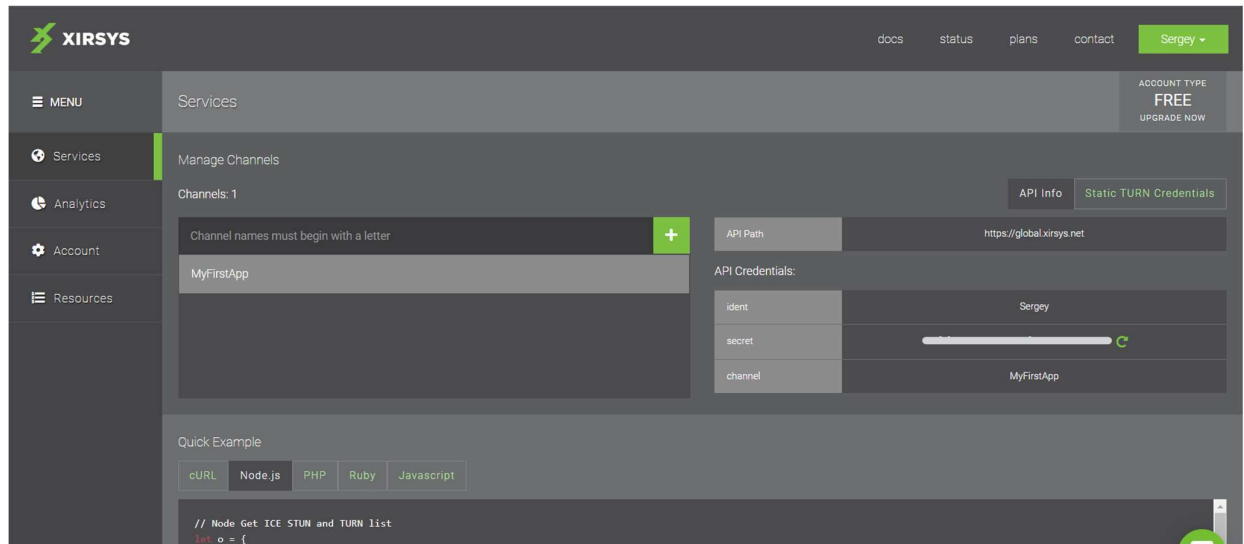
```
{"iceServers":[{"urls":"stun:stun.x.x.com:xxxx"}]}
```

Buttons: Cancel Save & New Save

Examples

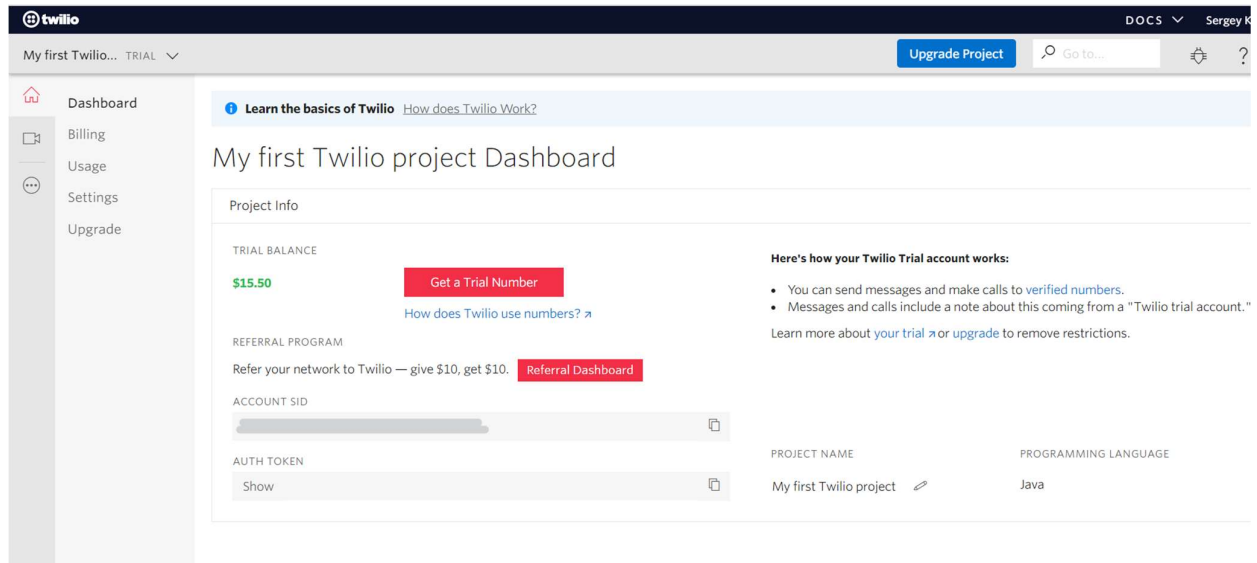
xirsys:

Navigate to Services tab (<https://global.xirsys.net/dashboard/services>) and here you can API Credentials, examples of requesting and even Static TURN Credentials, that you can paste to your custom settings as JSON file.



Twilio:

Navigate to your dashboard tab (<https://www.twilio.com/console>) and find your account credentials. Using them, you can request your "iceServers".



For example you can make POST request:

URL: https://api.twilio.com/2010-04-01/Accounts/ACCOUNT_SID/Tokens.json

Type: basic Auth

Username: **ACCOUNT SID;**

Password: **AUTH TOKEN;**

For more information read twilio documentation.

Documentation

Please, look at introduction to WebRTC protocols: https://developer.mozilla.org/en-US/docs/Web/API/WebRTC_API/Protocols

WebRTCConnection uses STUN servers to get public client address. Some routers using NAT employ a restriction called 'Symmetric NAT'. To bypass this restriction WebRTCConnection configuration has option to set TURN servers for relaying all information through that TURN server.

To have ability set STUN/TURN servers in WebRTCConnection configuration there is in Video Chat (Video_Chat__c) custom object record in *RTC Connection Settings* field.

As result SF system administrator is able to set STUN/TURN servers configuration:

```
{
  iceServers: [
    {
      urls: [ "stun:eu-turn4.xirsys.com" ]
    },
    {
      username: "enter username",
      credential: "enter credential",
      urls: [
        "turn:eu-turn4.xirsys.com:80?transport=udp",
        "turn:eu-turn4.xirsys.com:3478?transport=udp",
        "turn:eu-turn4.xirsys.com:80?transport=tcp",
        "turn:eu-turn4.xirsys.com:3478?transport=tcp",
        "turns:eu-turn4.xirsys.com:443?transport=tcp",
        "turns:eu-turn4.xirsys.com:5349?transport=tcp"
      ]
    }
  ]
}
```

1. STUN SETUP:

SF system administrator has to setup STUN configuration:

- use public STUN servers(has to check SLA). Format is next:{"iceServers":[{"urls":"stun:[stun1.l.XXXXX:XXXX](#)"},{urls":"stun:[stun1.l.XXXXX:19302](#)"},{urls":"stun:XXX:19302"}},{urls":"stun:[stun3.l.XXXXX:19302](#)"},{urls":"stun:[stun.XXXX.com:3478](#)"}]}. NOTE: some public STUNs don't have SLA.
- setup your own STUN server
- register in any STUN/TURN cloud providers and use their STUN server(for example, <https://xirsys.com/>)

2. TURN SETUP:

There aren't public TURN servers. As result to setup TURN configuration SF system administrator has to:

- setup your own TURN server
- register in any STUN/TURN cloud providers and use their TURN server(for example, <https://xirsys.com/>)

3. NOTE:

It's possible to use Video Chat Call only with STUN server configuration(without TURN servers). It will cover next type of connections with 1 NAT:

- between smartphone and computer (1 NAT: home or work LAN)
- between 2 computers in the same LAN

TURN server configuration is required to make call from different LANs (2 NATs)