

# How to combine Webitel and 3CX numbering plans

## Integration Webitel with PBX 3CX

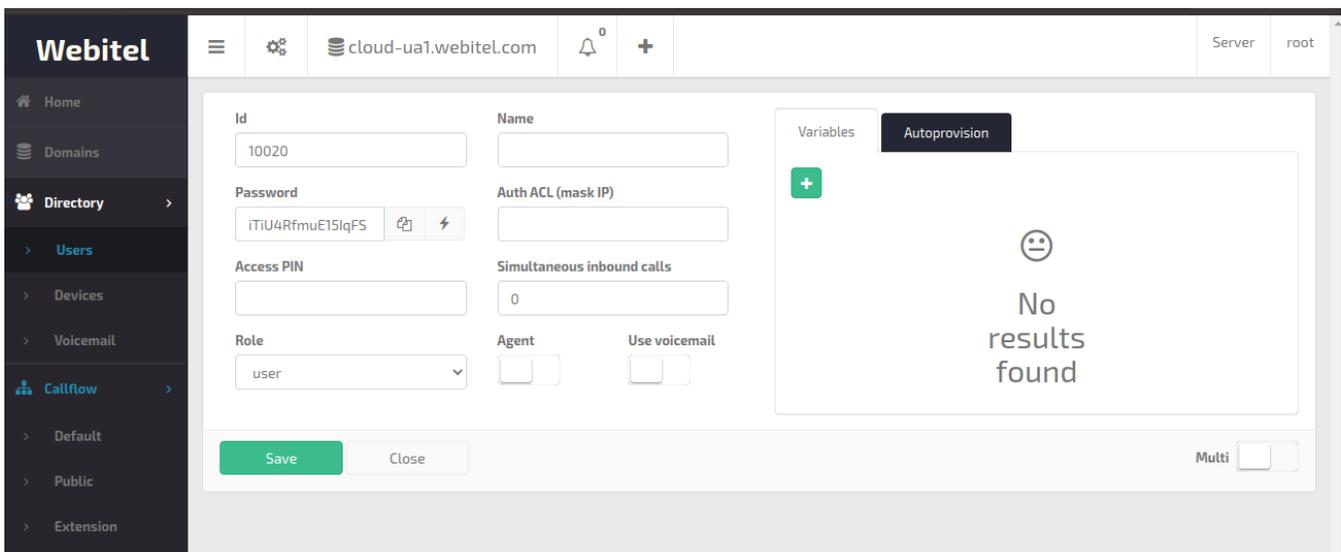
To combine numbering plans on the 3CX side, you need to create 2 SIP trunks - one for inbound calls, and the other for outbound calls.

In this example, the numbering plan on the 3CX side is 3-digit numbers (which start with "5"), and on the Webitel side - 4-digit numbers (which start with "1").

## Configuration example

## Calls from 3CX to Webitel

In Webitel, in the Users section, create a new user with the number 10020:

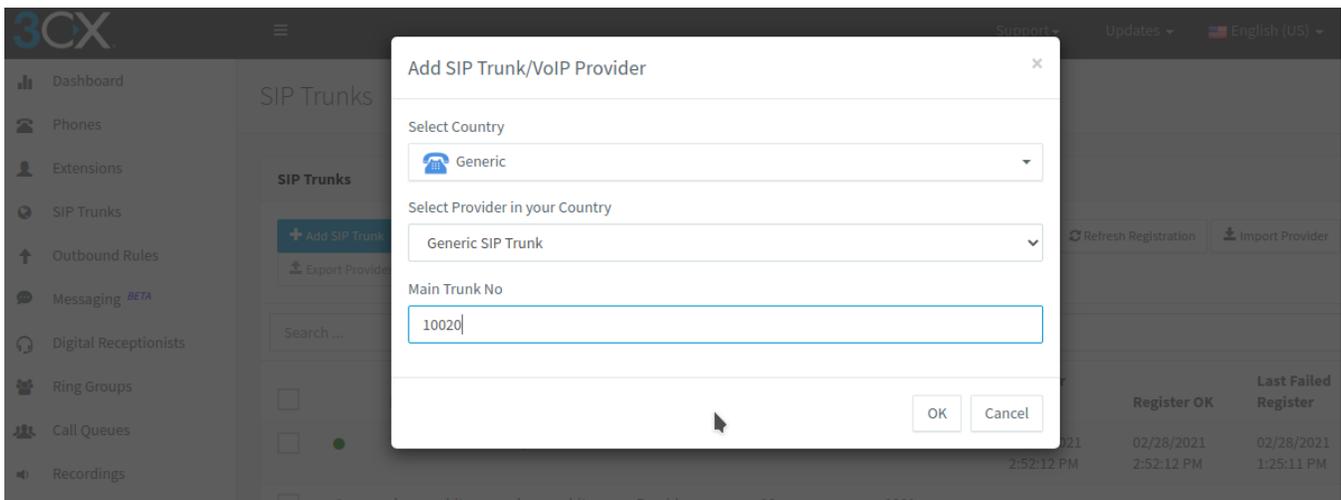


The screenshot shows the Webitel user creation interface. The left sidebar contains navigation options: Home, Domains, Directory, Users, Devices, Voicemail, Callflow, Default, Public, and Extension. The main form has the following fields:

- Id:** 10020
- Name:** (empty)
- Password:** iTiU4RfmuE15lqFS
- Auth ACL (mask IP):** (empty)
- Access PIN:** (empty)
- Simultaneous inbound calls:** 0
- Role:** user
- Agent:** (checkbox, unchecked)
- Use voicemail:** (checkbox, unchecked)

Buttons include "Save", "Close", and "Multi" (checkbox). A "Variables" section is visible with a "+" icon, and an "Autoprovision" tab is active. A message box on the right states "No results found".

On the 3CX sides, create a SIP Trunk with the "Generic" type:



The screenshot shows the 3CX "Add SIP Trunk/VoIP Provider" dialog box. The background shows the 3CX dashboard with a "SIP Trunks" list. The dialog box contains the following fields:

- Select Country:** Generic
- Select Provider in your Country:** Generic SIP Trunk
- Main Trunk No:** 10020

Buttons include "OK" and "Cancel".

In the gateway card, we write the registration data of the Webitel user: Username, password, domain, sip server:

# Generic SIP Trunk

OK

Cancel

Help

General DIDs Caller ID Options Inbound Parameters Outbound Parameters

## Trunk Details

Enter name for Trunk

Generic SIP Trunk

Registrar/Server/Gateway Hostname or IP

cloud-ua1.webitel.com

Auto Discovery

Outbound Proxy

sip-pstn.webitel.com

5070

Auto Discovery

Number of SIM Calls

10

## Authentication

Type of Authentication

Registrar/Account based

Authentication ID (aka SIP User ID)

10020

Authentication Password

\*\*\*\*\*

3 Way Authentication Password

After saving, the trunk should be registered successfully:

<input type="checkbox"/>	Name	Host	Type	Sim Calls	Main Trunk No	Register Sent	Register OK	Last Failed Register	<input type="checkbox"/>
<input type="checkbox"/>	● Webitel SIP Trunk	cloud-ua1.webitel...	Provider	10	10020	03/01/2021 9:25:01 AM	03/01/2021 9:25:01 AM	--	<input type="checkbox"/>

Go to the "Outbound Rules" menu on the 3CX side and create a rule that all 4-digit numbers should be sent in this trunk:

3CX Portal

Support ▾ Updates ▾ English (US) ▾ BK

Webitel 1XXX OK Cancel Help

**General**

Rule Name  
Webitel 1XXX

**Apply this rule to these calls**

Calls to numbers starting with prefix  
Calls to numbers starting with prefix

Calls from extension(s)  
Calls from extension(s)

Calls to Numbers with a length of  
4

Calls from extension group(s)  
+ Add

**Make outbound calls on**

Configure up to 5 backup routes for outgoing calls. Each route can be configured differently

Route		Strip Digits	Prepend	Outbound Caller ID
Route 1	Webitel SIP Trunk	0		
Route 2	BLOCK CALLS	0		
Route 3	BLOCK CALLS	0		
Route 4	BLOCK CALLS	0		
Route 5	BLOCK CALLS	0		

Now we can make calls from 3CX to any Webitel internal numbers.

## Calls from Webitel to 3CX

In 3CX, in the SIP Trunks section, add a new entry with the "Bridge - Slave" type:

Webitel slave Incoming OK Cancel Help

General Presence Advanced

**General**

Name of bridge

Virtual extension number

Outbound rule prefix to reach remote 3CX PBX

Maximum Simultaneous Calls

**Authentication**

This "Master" Bridge will receive registrations only. The other PBX must register with this password:

👁

We give the name, register the login 10021 and create a password.

In the Gateways section on the Webitel side, create a new gateway with registration, where we enter the data of the newly created bridge:

**Webitel** voicemail cloud-ua1.webitel.com 0 + Server root

**Callflow** Default Public Extension Blacklists Variables Queue **Gateways** Calendar Media

Name

Type

General Parameters

Host/IP

User Name

Password  
 👁

Save Cancel Close

After saving, successful registration should occur:

OmniLine External 👇 🟢 REGED ↑ sip:1001

Go to the "Default callflow" section and create a routing scheme in which we indicate that all 2-digit numbers that start with "5" must be sent through this gateway:

The screenshot shows the Webitel interface for configuring a gateway. The top navigation bar includes the Webitel logo, a menu icon, a settings icon, the domain 'cloud-ua1.webitel.com', a notification bell with '0', and a plus sign. The left sidebar contains navigation options: Home, Domains, Directory, Callflow (selected), and its sub-items: Default, Public, Extension, Blacklists, Variables, Queue, Gateways, Calendar, and Media. The main content area is titled '3CXOut' and shows the 'Number' field with the regular expression '^5\d{2}\$' and the 'Time zone' set to 'Europe/Moscow'. Below this, the 'Callflow' section is active, showing a configuration for 'On disconnect'. The configuration is a JSON object with the following structure:

```
1 [
2
3 {
4   "ringback": {
5     "call": {
6       "name": "${ru-ring}",
7       "type": "tone"
8     }
9   },
10  "transfer": {
11    "name": "${ru-ring}",
12    "type": "tone"
13  }
14 }
15 {
16   "recordSession": {
17     "action": "start",
18     "type": "mp3",
19     "stereo": true,
20     "followTransfer": true,
21     "bridged": true,
22     "minSec": 2,
23     "email": []
24   }
25 }
26 {
27   "bridge": {
28     "endpoints": []
```

```
[
  {
    "ringback": {
      "call": {
        "name": "${ru-ring}",
        "type": "tone"
      },
      "transfer": {
        "name": "${ru-ring}",
        "type": "tone"
      }
    }
  },
  {
    "recordSession": {
      "action": "start",
      "type": "mp3",
      "stereo": true,
      "followTransfer": true,
      "bridged": true,
      "minSec": 2,
      "email": []
    }
  },
  {
    "bridge": {
      "endpoints": [
        {
          "type": "sipGateway",
          "name": "3CX",
          "dialString": "$1"
        }
      ]
    }
  },
  {
    "hangup": "NORMAL_CLEARING"
  }
]
```

Now we can make calls from Webitel to any 3CX internal numbers.

 The description is relevant for Webitel version 3.11