fonio

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3CX

SIP Trunk Integration Guide

The complete, step-by-step guide to connecting your 3CX PBX to Fonio AI, written for users of all experience levels

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Document Version	2.0 - Complete Rewrite
Covers	3CX V20 (also applicable to V18)
Estimated Setup Time	20–35 minutes for first-time users
Support	support@fonio.ai

Before You Start - Read This First

This guide explains every single step required to connect your 3CX phone system to Fonio's AI voice platform. Read this section before touching any settings. Understanding what you are building before you build it will save you significant time.

What You Are Actually Building

Fonio is an AI voice assistant that answers phone calls. For Fonio to answer calls that come through your 3CX phone system, 3CX needs a direct SIP connection to Fonio's platform. This connection is called a SIP Trunk.

Think of it like a private phone line between 3CX and Fonio. When a call arrives at your 3CX system and you want Fonio to handle it, 3CX forwards that call down this private line to Fonio. Fonio's AI answers, handles the conversation, and can transfer the caller back to a human on 3CX if needed.

How the call travels - the complete path

Step 1: A caller dials your phone number from their phone

Step 2: Your phone carrier (Telnyx, Twilio, etc.) or 3CX's own trunk delivers the call into 3CX

Step 3: the outbound rule matches the call and routes it to the Fonio trunk.

Step 4: 3CX forwards the call through the Fonio SIP Trunk to Fonio's servers

Step 5: Fonio's AI assistant picks up and handles the conversation

Step 6 (if needed): Fonio transfers the caller back to a human agent on 3CX, the call returns through the same trunk

Requirements - Check These Before Starting

You must meet ALL of the following requirements. If you miss even one, the setup will not work.

3CX Requirements

License: You need a 3CX Business, PRO, or Enterprise license. The Free plan and SMB plan do NOT support Generic SIP Trunk (IP Based), the specific trunk type this guide uses. If you are on Free or SMB, you will not see the option and cannot proceed until you upgrade.

Deployment type: Your 3CX must be either Self-Hosted (on your own server or VPS) or on a 3CX Dedicated Hosted instance. The standard 3CX-managed hosted plans (yourname.3cx.eu) do not support this trunk type.

Version: 3CX V20 (recommended) or V18. Screenshots in this guide show V20.

Admin access: You must be able to log in to the 3CX Admin Console, the cog icon at the bottom-left of the 3CX interface. Regular user access is not enough.

Public IP: Your 3CX server must have a reachable public IP address. Fonio will send voice traffic back to this IP. If your server sits behind NAT with no public IP, configure STUN (see Step 1).

A working extension: You need at least one registered 3CX extension (softphone, IP phone, or 3CX web client) to test the connection at the end.

Fonio Requirements

Plan: You need the Fonio Team plan or higher. The Solo plan does NOT include the 'Own SIP Trunk' feature that this guide requires. Log in to Fonio and check your current plan before starting.

A configured assistant: You need at least one Fonio AI assistant already set up with a voice, a language, and a start message (what the AI says when it answers a call). If you have not done this yet, set up your assistant first, then return to this guide.

Quick Compatibility Check

Before spending 30 minutes on setup, do these two checks right now:

Check 1 - 3CX: Log in to your 3CX Admin Console. Click the Admin Cog (bottom left) → Voice & Chat. Click 'Add SIP Trunk' → 'Add Trunk'. On the trunk creation page, set Country to 'Worldwide'. Look in the Provider dropdown, do you see 'Generic SIP Trunk (IP Based)'? If YES, you can proceed. If NO, your plan does not support it and you need to upgrade before continuing.

Check 2 - Fonio: Log in to your Fonio account. Click 'Phone Numbers' in the left sidebar. Click 'Add Number'. Do you see an option called 'Create SIP Number' or 'SIP Nummer erstellen'? If YES, you can proceed. If NO, you are on the Solo plan and need to upgrade.

Technical Details - What You Do Not Need to Memorise

You do not need to understand these details to complete the setup. They are here for reference and for troubleshooting. Fonio's SIP infrastructure runs on LiveKit Cloud. Here are the key technical facts:

Parameter	Value
Fonio SIP Server (Registrar)	`vojhwtf1hn6.eu.sip.livekit.cloud`
SIP Port	`5060`
SIP Transport	TCP only (NOT UDP - this is critical)
Authentication Method	IP-based - no username or password
Fonio IP Range 1	`143.223.88.0/21`
Fonio IP Range 2	`161.115.160.0/19`

Voice Audio (RTP) Ports	UDP 10000–20000 (must remain open even though SIP uses TCP)

Why TCP for SIP and UDP for audio?

SIP signaling (the messages that set up and tear down calls) must use TCP to connect to Fonio's LiveKit servers reliably. If you leave it on UDP, calls will connect intermittently, drop suddenly, or have no audio.

However, the actual voice audio (called RTP) always travels over UDP, this is standard for all VoIP systems. You must NOT block UDP, even though you are setting SIP to TCP. These are two completely different traffic types. One rule: SIP = TCP. Audio = UDP.

PHASE 1

Fonio Configuration

Create your SIP number in Fonio first, you will need it before configuring 3CX

Complete everything in Phase 1 inside your Fonio account before touching 3CX. You need two pieces of information from Fonio before 3CX configuration begins: (1) the SIP number you create, and (2) the Fonio server address. Both are covered in this phase.

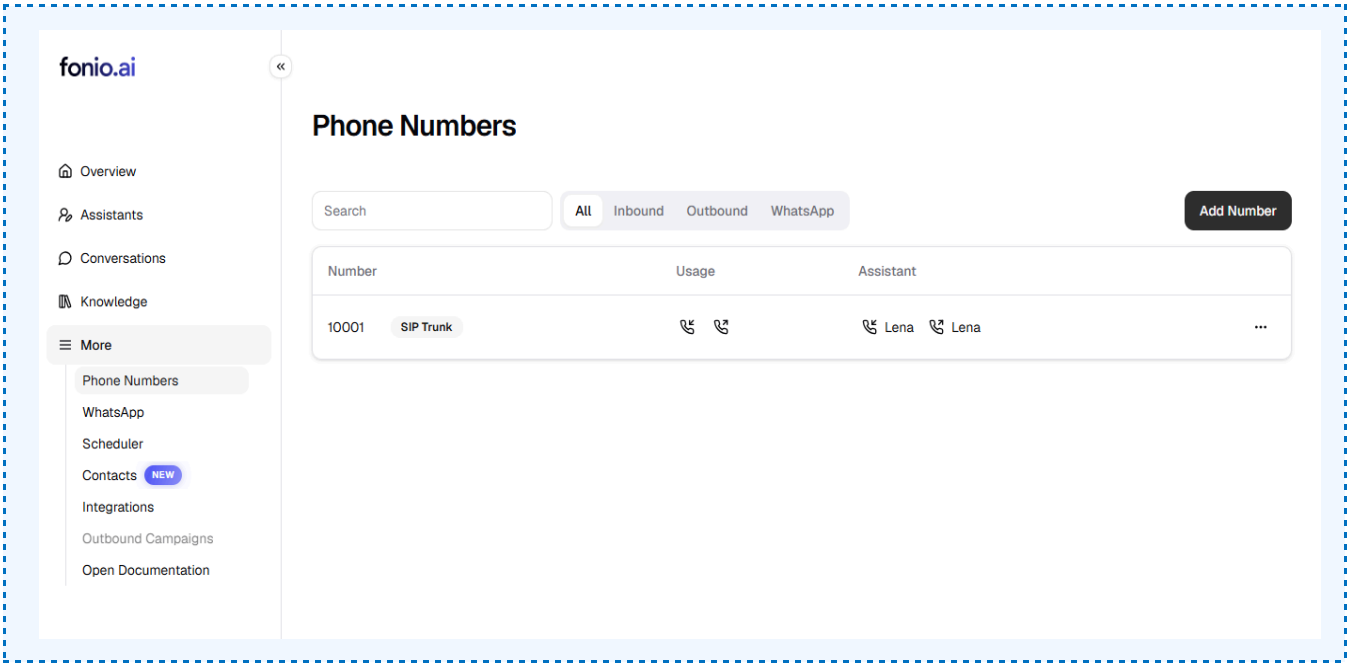
Step 1 Log in to Fonio and confirm your plan

Go to app.fonio.ai and sign in with your email and password. Once you are in the dashboard, navigate to your account settings and confirm you are on the Team plan or higher.

> ⚠️ **If you are on the Solo plan**
>
> The 'Create SIP Number' option will not appear. You must upgrade your Fonio plan to Team before continuing. Go to Settings → Plan & Billing to upgrade.

Step 2 Open the Phone Numbers section

In the left sidebar of the Fonio dashboard, click Phone Numbers. You will see a list of any numbers you currently have. Click the Add Number button at the top right corner.



Step 3 Select 'Create SIP Number'

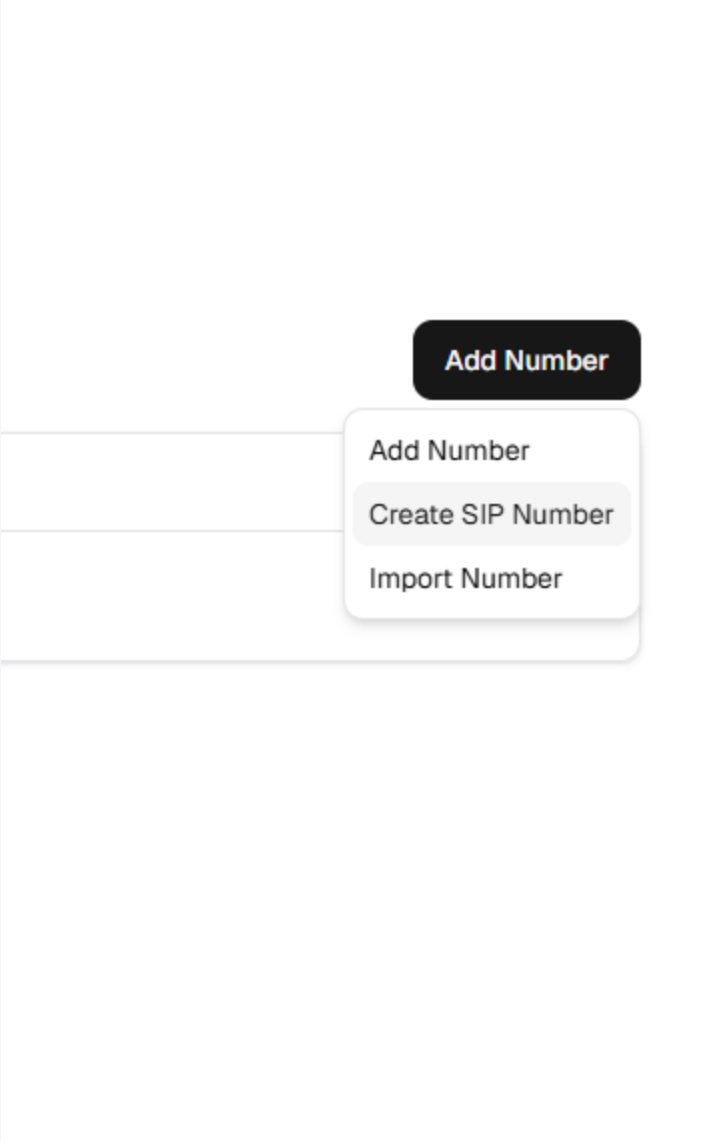
After clicking Add Number, a dialog box or menu appears with options for what type of number to add. Select Create SIP Number.

ℹ️ What is a SIP Number and why is it different from a regular number?

A regular phone number (DID) lets people call you from the public phone network, it is a number like +1 212 555 0100 that anyone can dial.

A SIP Number is completely different. It is an internal routing identifier, a simple numeric code (like 9000) that only exists within the SIP layer between 3CX and Fonio. It is not a phone number anyone can call. It is the 'address' that 3CX uses to reach Fonio. Nobody outside your system ever sees or uses this number.

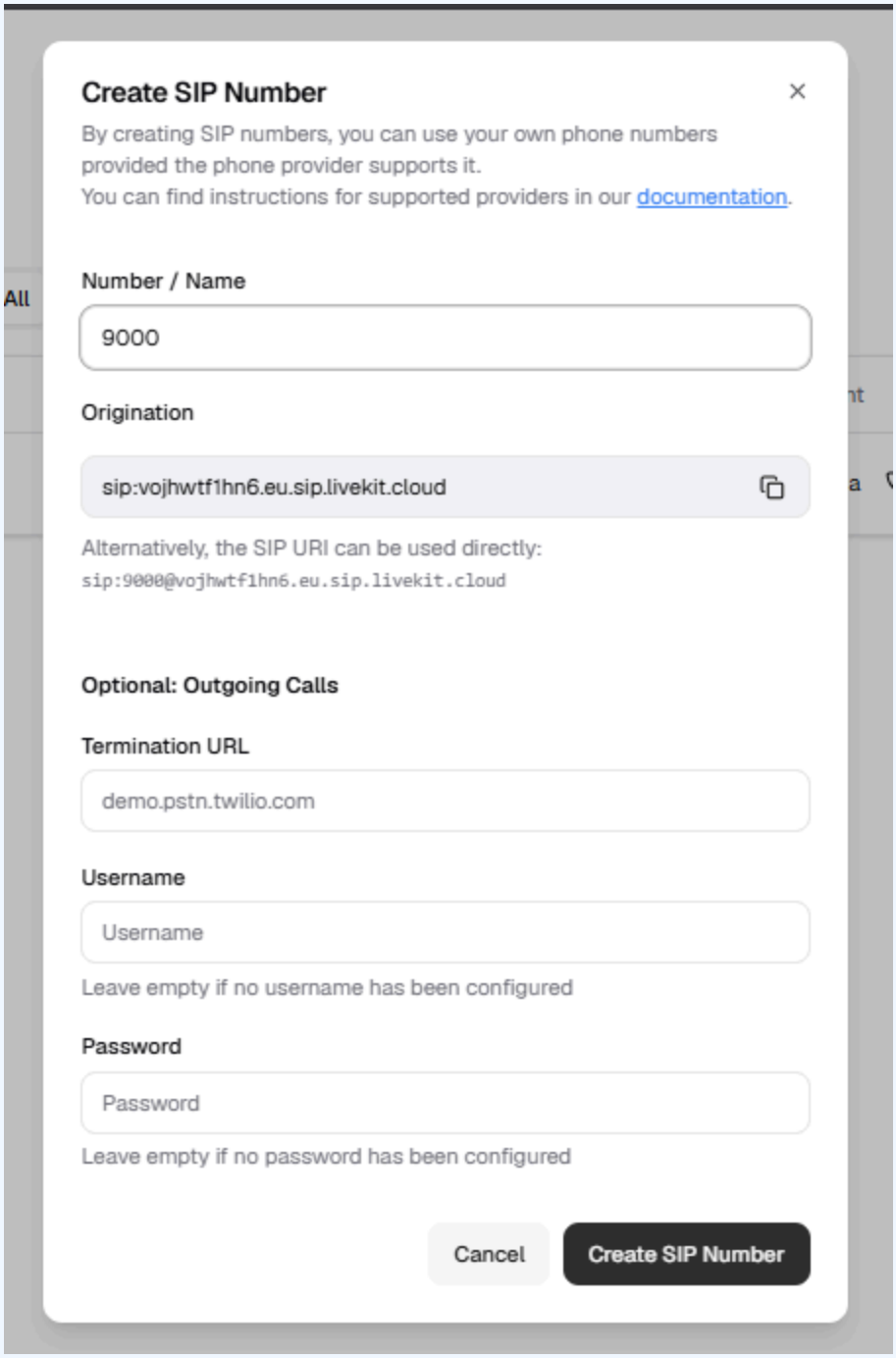
Think of it like an internal office extension. Dialing extension 9000 from inside the office connects you to Fonio. But 9000 means nothing outside the building.



Step 4 Enter your SIP number

A field appears asking you to enter the SIP number. Type a short, simple numeric value. Do not add any symbols, spaces, or a + sign, just plain digits.

> ⚠️ **If Fonio says the number is already in use**
>
> Fonio's SIP numbers are shared across all customers worldwide. If 9000 is taken, it means another Fonio customer claimed it first. Simply try a different number. The number does not need to look like a real phone number, any unique short numeric string works.
>
> **Important:** Write down the exact number you successfully create. You will enter this same number in 3CX later. If they do not match exactly, the connection will fail silently.



Step 5 Save the SIP number

Click Create or Save. The number will appear in your Phone Numbers list tagged as a SIP Trunk. The trunk has not been connected yet.

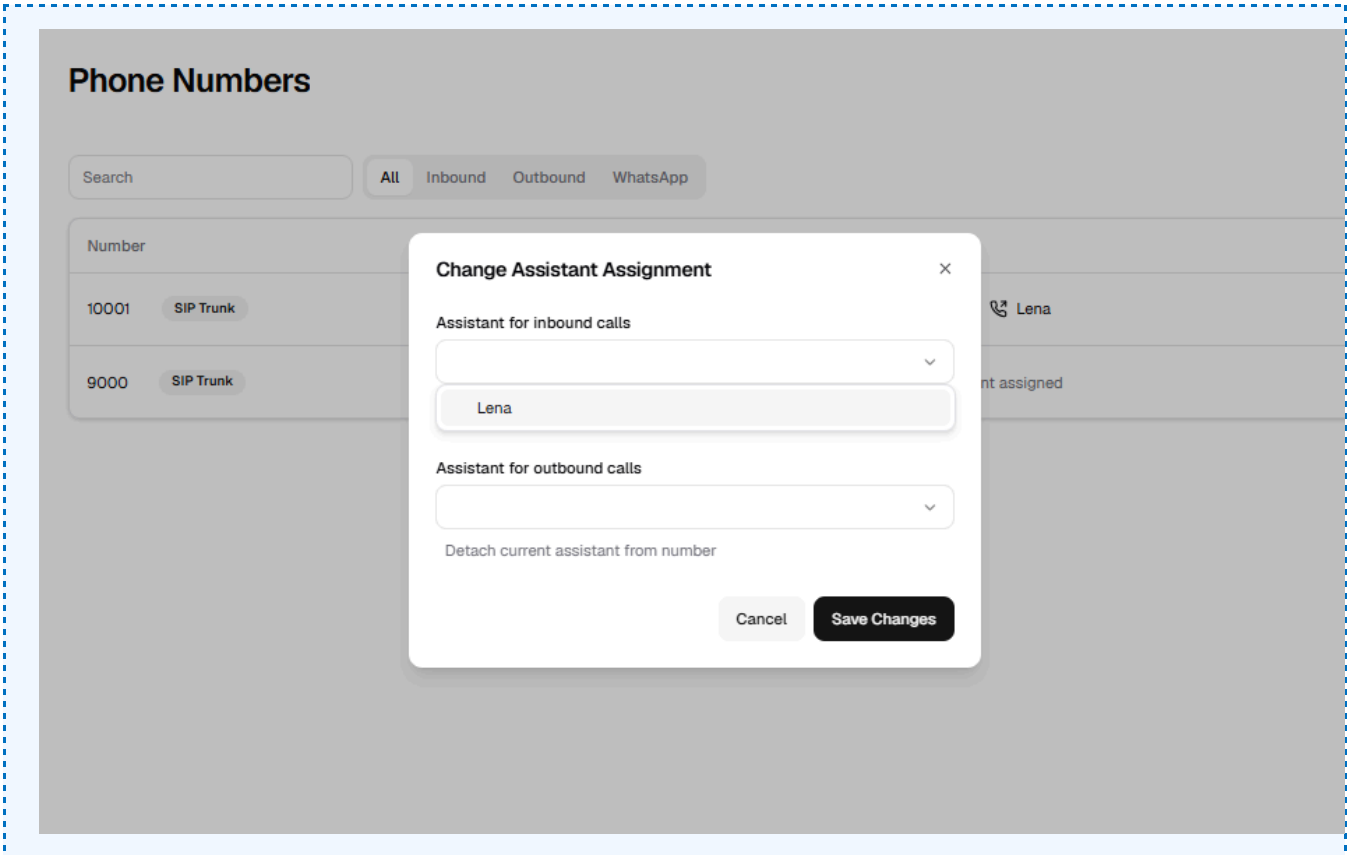
Step 6 Open your Fonio assistant

In the Fonio left sidebar, navigate to your AI assistant. If you have multiple assistants, select the one that should answer calls coming through 3CX.

> ⚠️ **If you have not created a Fonio assistant yet**
>
> You must create one before continuing. Click Assistants (or the equivalent) in the left sidebar → Create New Assistant. You need to configure at minimum:
>
> - **Voice and Language:** Choose a voice and set the language (e.g. English)
> - **Start Message:** The first thing the AI says when it answers. Example: 'Hello, thank you for calling. How can I help you today?' Without a start message the AI will answer in silence.
>
> Once your assistant is configured, return here.

Step 7 Assign the SIP number to the assistant

Inside your assistant settings, find the Phone Numbers or Number Assignment section. Click Assign Number (or Nummer zuweisen). From the dropdown, select the SIP number you created in Step 4 (e.g. 9000). Click Save.



The SIP number is now linked to your assistant. When Fonio receives a call destined for SIP number 9000, it will route that call to this specific assistant.

✅ Phase 1 Complete - Fonio is Ready

You have completed everything needed on the Fonio side. Here is a summary of what you now have:

- **Your Fonio SIP number:** the number you created in Step 4
- **Fonio SIP server:** `vojhwtf1hn6.eu.sip.livekit.cloud` port `5060` transport: TCP
- **Your assistant:** Assigned to the SIP number above, with a voice and start message configured

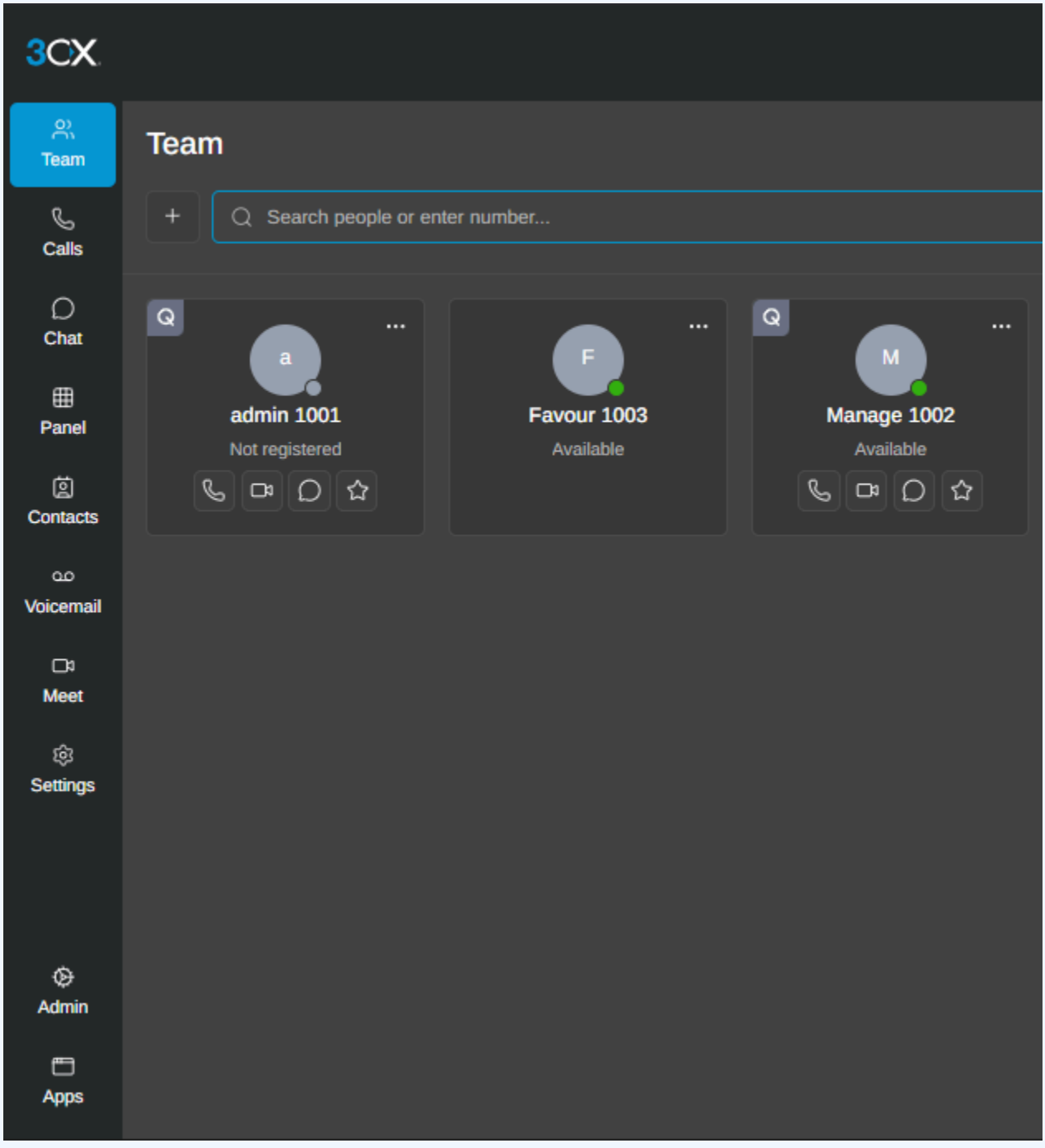
Do not change anything in Fonio for the rest of this guide. All remaining steps are inside 3CX.

PHASE 2

3CX Configuration

Four tasks: network check, IP whitelist, create the Fonio trunk, set up routing rules

Everything in Phase 2 takes place inside the 3CX Admin Console. To access it, open your 3CX client (web browser or desktop app) and click the Admin Cog icon at the bottom-left of the interface.



You will now complete four tasks inside 3CX, in this exact order:

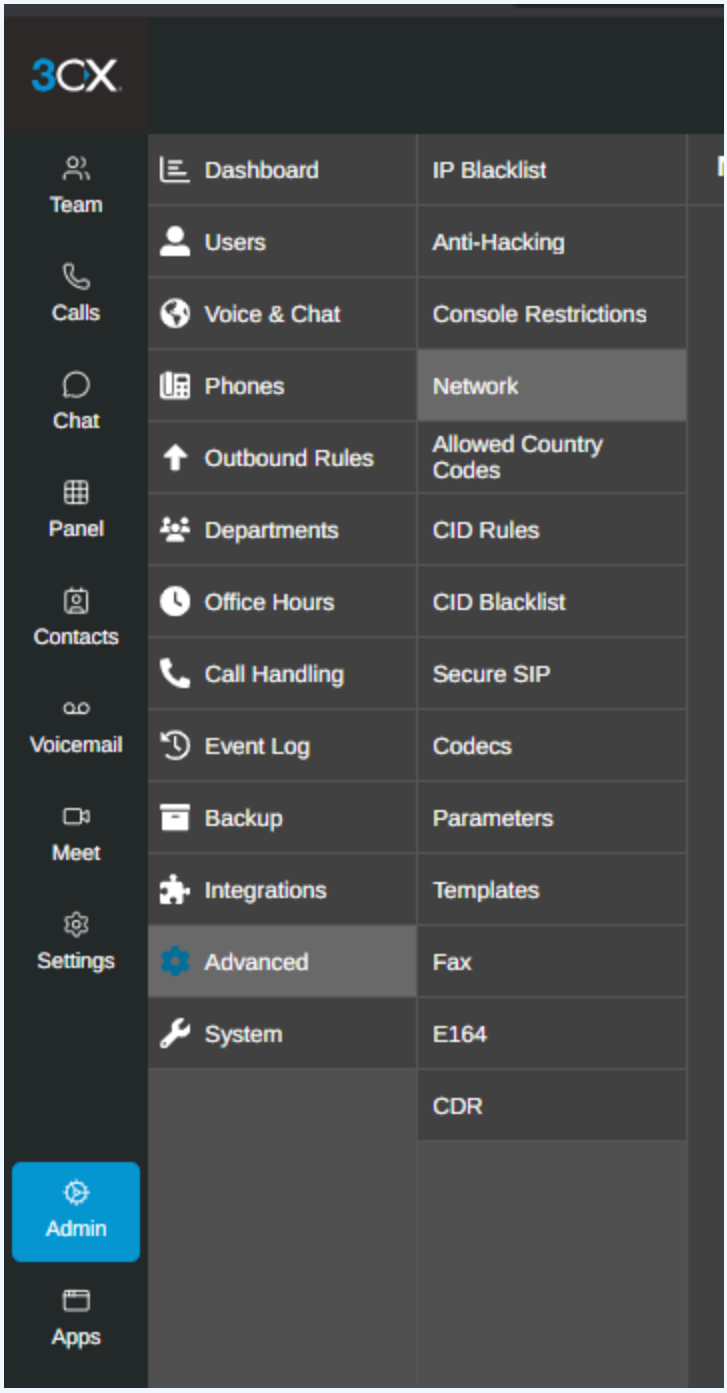
1. Check and confirm your network / external IP settings
2. Whitelist Fonio's IP addresses so their traffic can reach your 3CX
3. Create the Fonio SIP Trunk
4. Create the routing rule (Outbound Rule)

Task 1 - Verify Network and External IP Settings

This is the most commonly skipped step and one of the most impactful. Fonio will send voice traffic back to your 3CX using your server's public IP address. If 3CX is reporting the wrong IP, Fonio cannot reach it and all calls will fail silently.

Step 8 **Open the Network settings**

In the 3CX Admin Console, click: Admin Cog → Advanced → Network tab.



Step 9 Confirm your External IP configuration

In the External IP Configuration section, you will see options for how 3CX announces its public IP address to external SIP systems (including Fonio). Choose the option that matches your setup:

Your situation	What to select

Your 3CX server has a fixed, permanent public IP address	Static IP, enter your public IP address in the field provided
Your server's IP address changes (dynamic IP)	Dynamic IP with STUN, 3CX will use a STUN server to automatically discover its current public IP. The default STUN server 3CX provides works well; you do not need to change it.
You are on 3CX Dedicated Hosted (3CX's own cloud)	This is pre-configured by 3CX. You can skip this step entirely.

> ⚠️ **How to check if this is set correctly**
>
> Open a browser on a separate device and go to whatismyip.com. Note the public IP address shown. Compare it to what is configured in 3CX's External IP field. They must match. If 3CX shows a private address like 192.168.x.x or 10.x.x.x, it is not correctly configured, Fonio will try to send audio to that private address and the call will connect in silence.

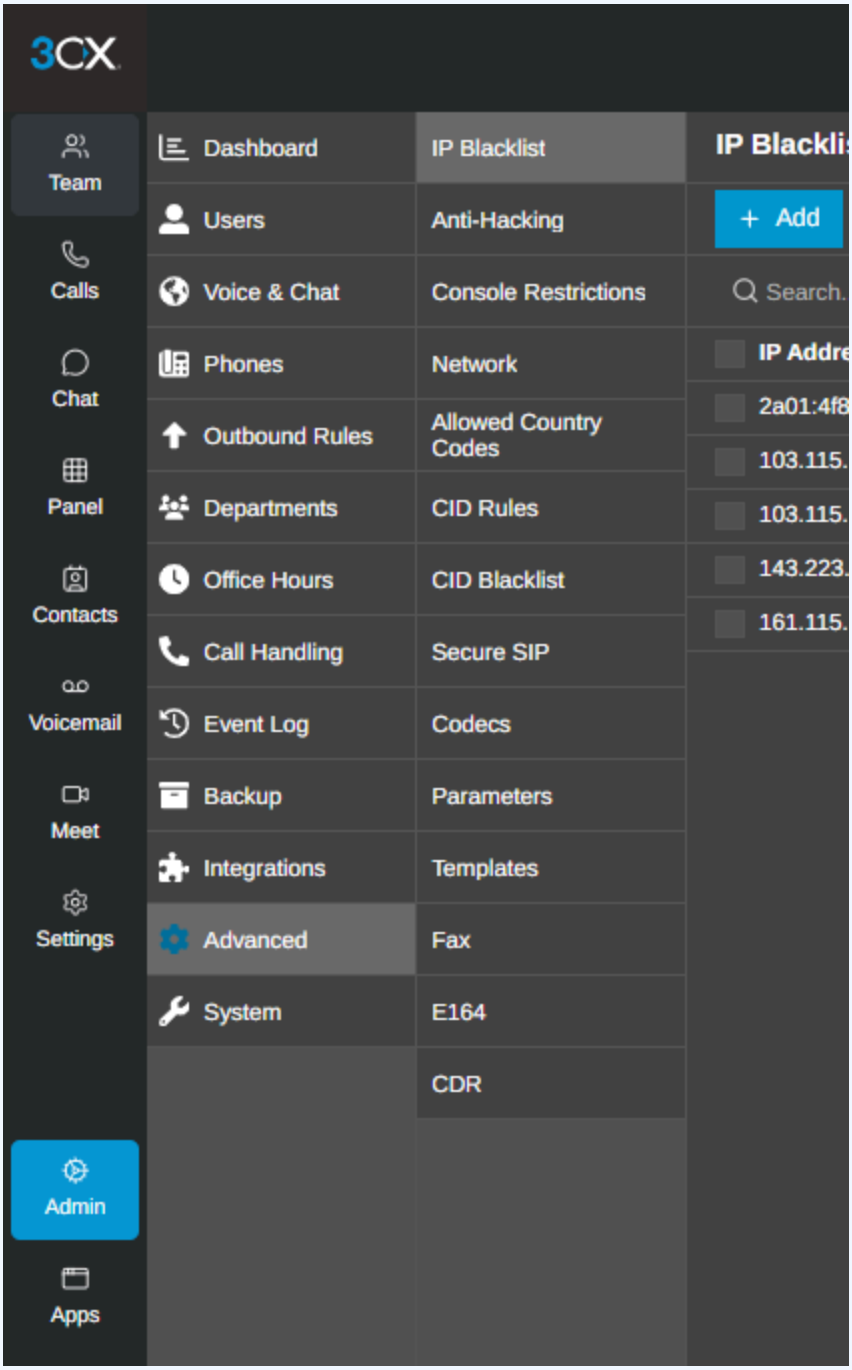
Task 2 - Whitelist Fonio's IP Ranges

3CX has a built-in firewall called the IP Blacklist. Despite the name, this is also where you ADD IP addresses that you want to ALLOW through. You must add Fonio's IP ranges here so that 3CX accepts SIP and audio traffic from Fonio's servers.

> **Why is it called 'IP Blacklist' if we are adding allowed IPs?**
>
> The name is confusing, this is a known issue with 3CX's interface. The IP Blacklist tab shows IPs that 3CX has automatically blocked (usually because they looked suspicious). By adding an IP range here with 'Allowed' status, you are telling 3CX to always accept traffic from those addresses, bypassing any automatic blocking. It is effectively an allowlist.

Step 10 **Navigate to the IP Blacklist**

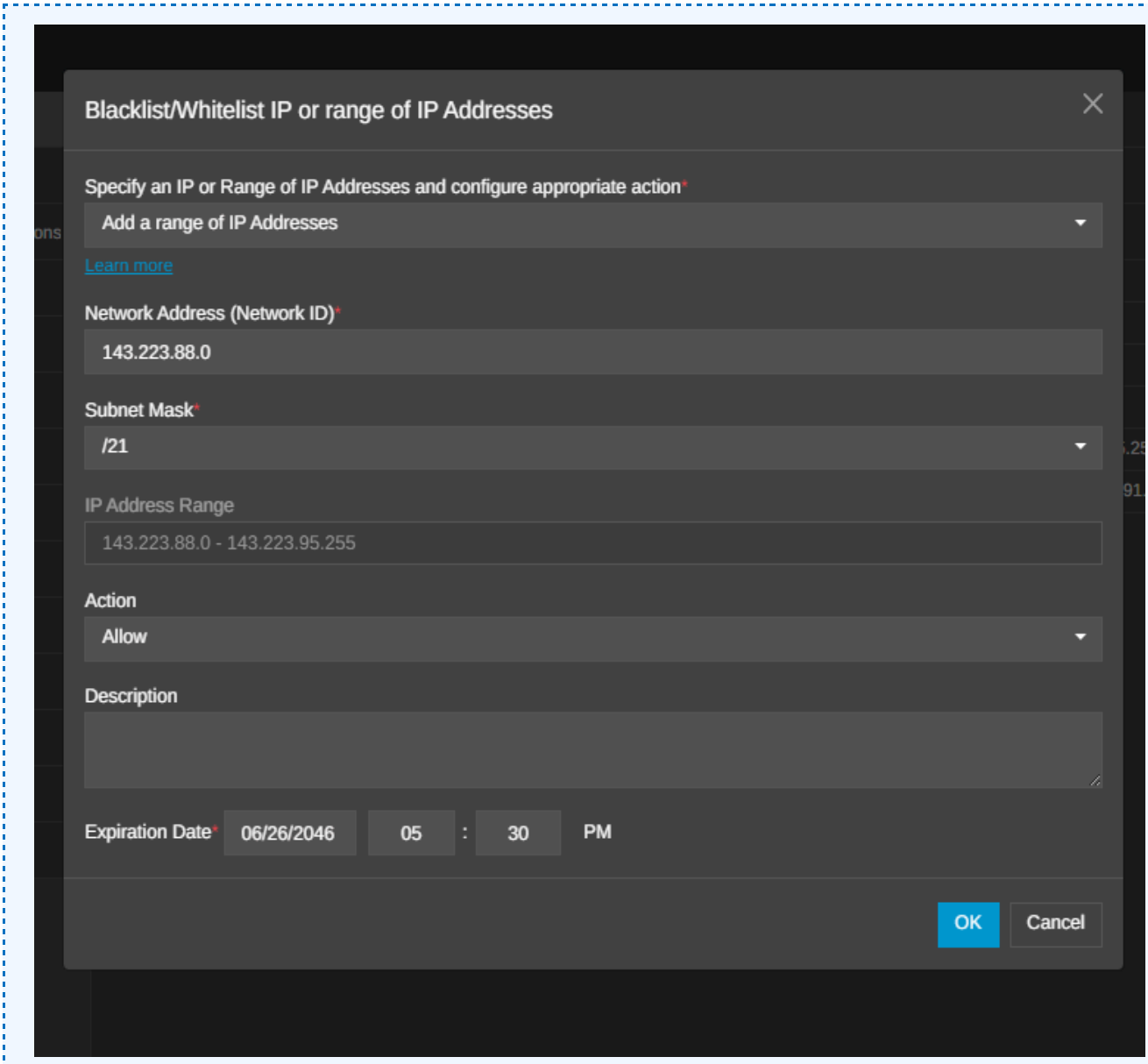
In the 3CX Admin Console: Admin Cog → Advanced → IP Blacklist tab.



Step 11 **Add the first Fonio IP range**

Click the **Add** button. In the IP address field, enter exactly: `143.223.88.0/21`

Set the Action to Allow (not Deny). Click Save or OK.

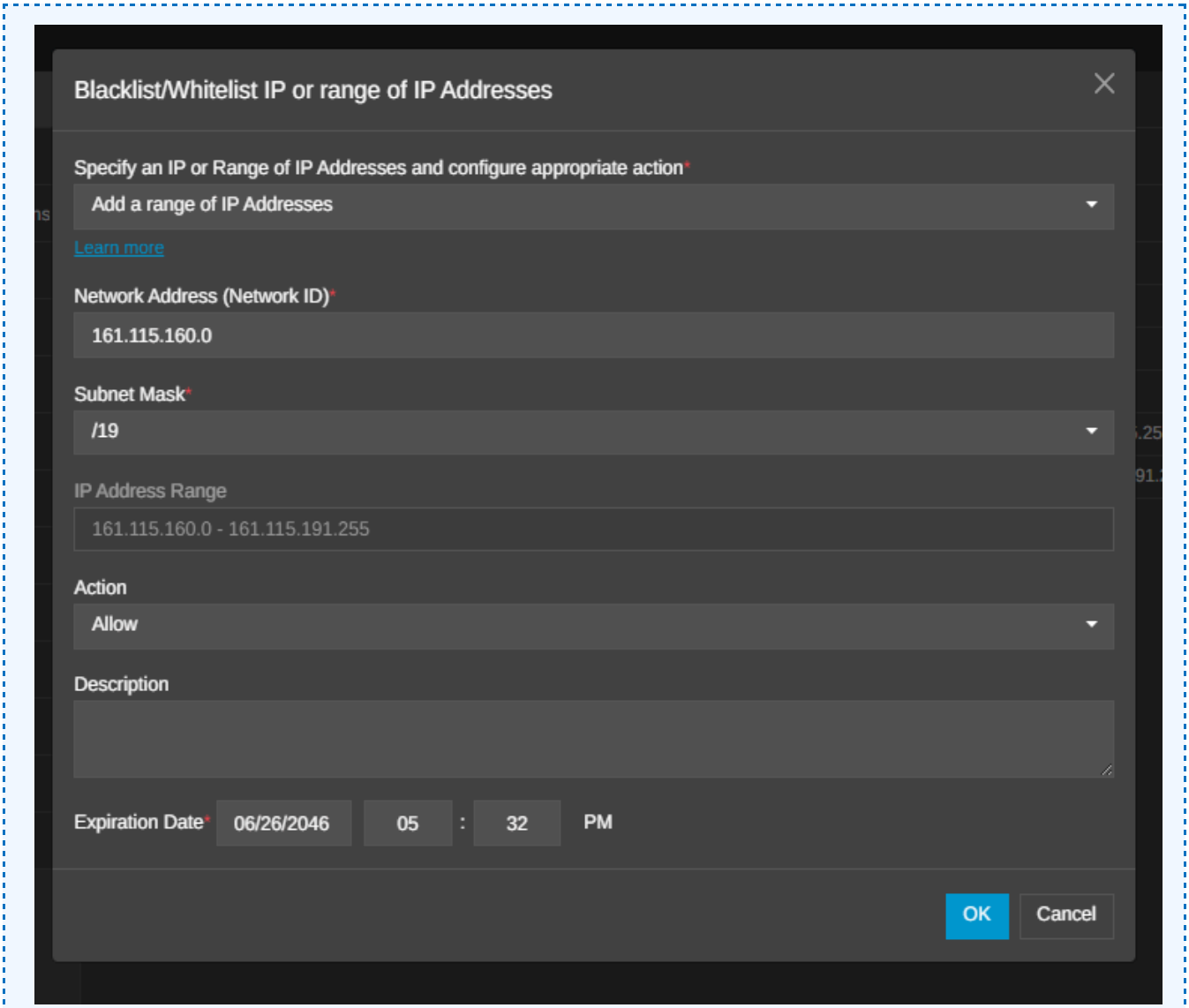


Step 12 **Add the second Fonio IP range**

Click **Add** again. Enter: `161.115.160.0/19`

Set the Action to Allow. Click Save or OK.

Both ranges must be present. Your IP Blacklist should now show both entries as Allowed.



⚠️ **If you are on a Self-Hosted 3CX - also open these ports at your server firewall**

Adding IPs to 3CX's internal IP Blacklist is not enough if your server also has an OS-level firewall (which all Linux/Windows servers have by default) or a cloud provider firewall (AWS Security Groups, Google Cloud Firewall, DigitalOcean Firewall, etc.).

You must also open these rules at your server or cloud firewall:

- `TCP port 5060` - for SIP signaling (both inbound FROM Fonio and outbound TO Fonio)
- `UDP ports 10000-20000` - for voice audio (RTP). This must be open in BOTH directions

If you are on 3CX Dedicated Hosted, 3CX manages the firewall for you and this step is not needed.

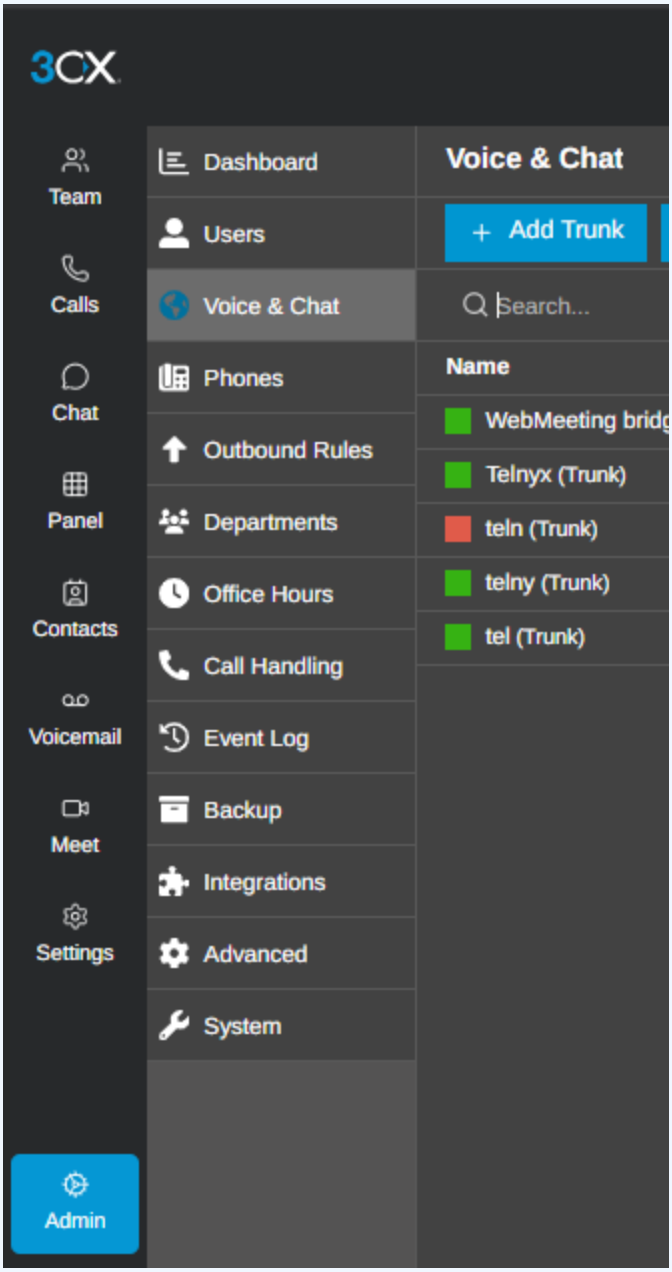
Task 3 - Create the Fonio SIP Trunk

This is the core of the setup. You are creating a direct SIP connection from 3CX to Fonio. This trunk uses IP-based authentication, no username or password, because Fonio identifies your 3CX by its public IP address.

> ℹ️ **What 'IP Based' trunk means - why it is different from your carrier trunk**
>
> If you already have a phone carrier trunk in 3CX (e.g. for Telnyx, Twilio, or another provider), that trunk registers with a username and password. You can see a green dot when it is connected.
>
> The Fonio trunk works differently. It does NOT register. It does NOT show a green dot. It has no username or password. Instead, 3CX simply sends calls directly to Fonio's IP address, and Fonio accepts them because it recognizes your 3CX's public IP. The trunk will show 'Idle' in 3CX, this is correct and expected, NOT an error.

Step 13 **Navigate to Voice & Chat and start a new trunk**

In the 3CX Admin Console, click: **Admin Cog** → **Voice & Chat**. You will see a list of any existing trunks. Click the **Add Trunk button**.



Step 14 **Select the Generic SIP Trunk type**

A trunk creation dialog or page opens. You will see two dropdowns: Country and Provider.

Country: Select your **Country or Worldwide**.

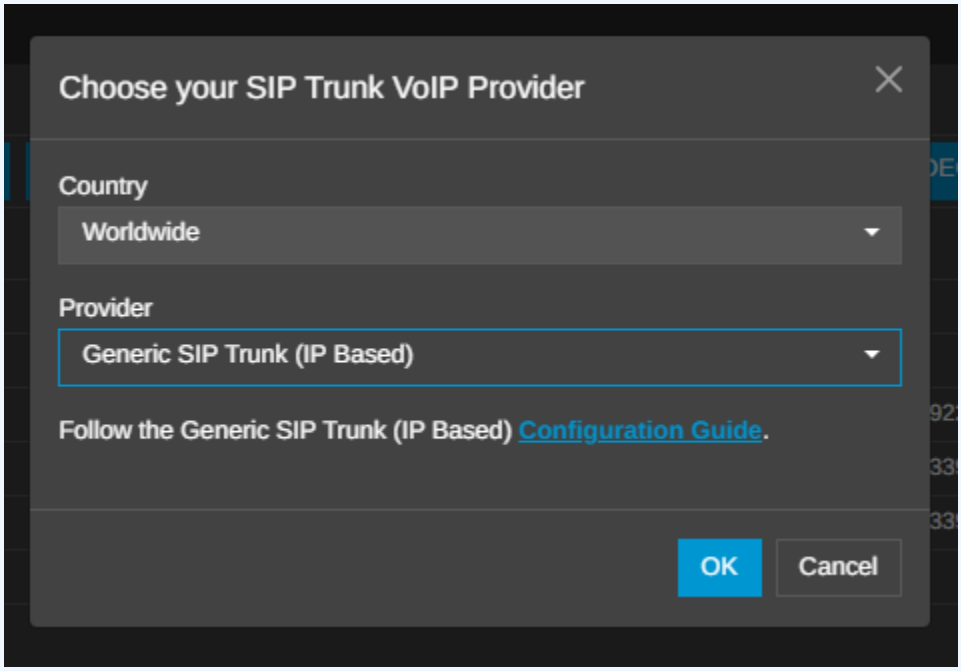
Provider: After making your selection, the Provider dropdown updates. Select **Generic SIP Trunk (IP Based)**.

⚠️ **CRITICAL - you must select 'Generic SIP Trunk (IP Based)' NOT 'Generic VoIP Provider (Registration)'**

Two similar-looking options exist. They behave completely differently:

- **Generic SIP Trunk (IP Based):** Correct for Fonio. No registration. Calls go directly to Fonio's IP. No username/password.
- **Generic VoIP Provider (Registration):** WRONG for Fonio. This tries to register with a username/password. Fonio's servers do not accept SIP registrations from PBXes. The registration will either fail or appear to work but calls will not route. This causes silent, hard-to-diagnose failures.

Always select IP Based - not Registration.



Step 15 Fill in the trunk General settings - Part 1: Name and Number

After selecting the trunk type, the main trunk configuration page opens. Fill in the fields exactly as described below.

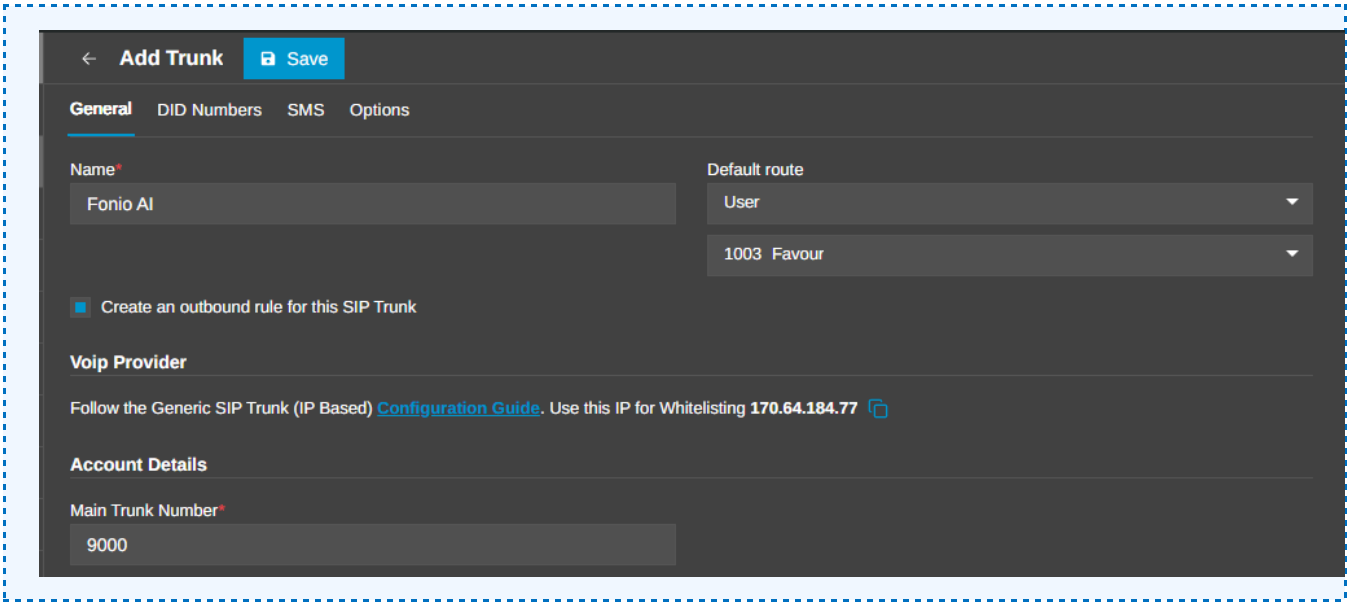
Field	What to Enter
Trunk Name	Fonio AI (or any name you will recognise - this is just a label)
Default Route	Set to your main extension or ring group (e.g. extension 100). This is where calls go if no specific inbound rule matches. You can change this later.

Main Trunk Number	Enter your Fonio SIP number, the one you created in Step 4 of Phase 1. Example: 9000. Use only digits, no + sign, no spaces, no country code.

The Main Trunk Number must be IDENTICAL to your Fonio SIP number

If you created SIP number `9000` in Fonio, enter exactly `9000` here. Not `+9000`. Not `09000`. Not `9000 ` (with a trailing space). Exactly `9000`.

When 3CX sends a call to Fonio, it uses this number as the destination in the SIP message. Fonio looks up this number to find which assistant should answer. If the number does not match exactly, even one character difference, Fonio cannot route the call and it will fail with no clear error message.



Step 16 — Fill in the trunk General settings - Part 2: Registrar and Authentication

Continue filling in the same General tab:

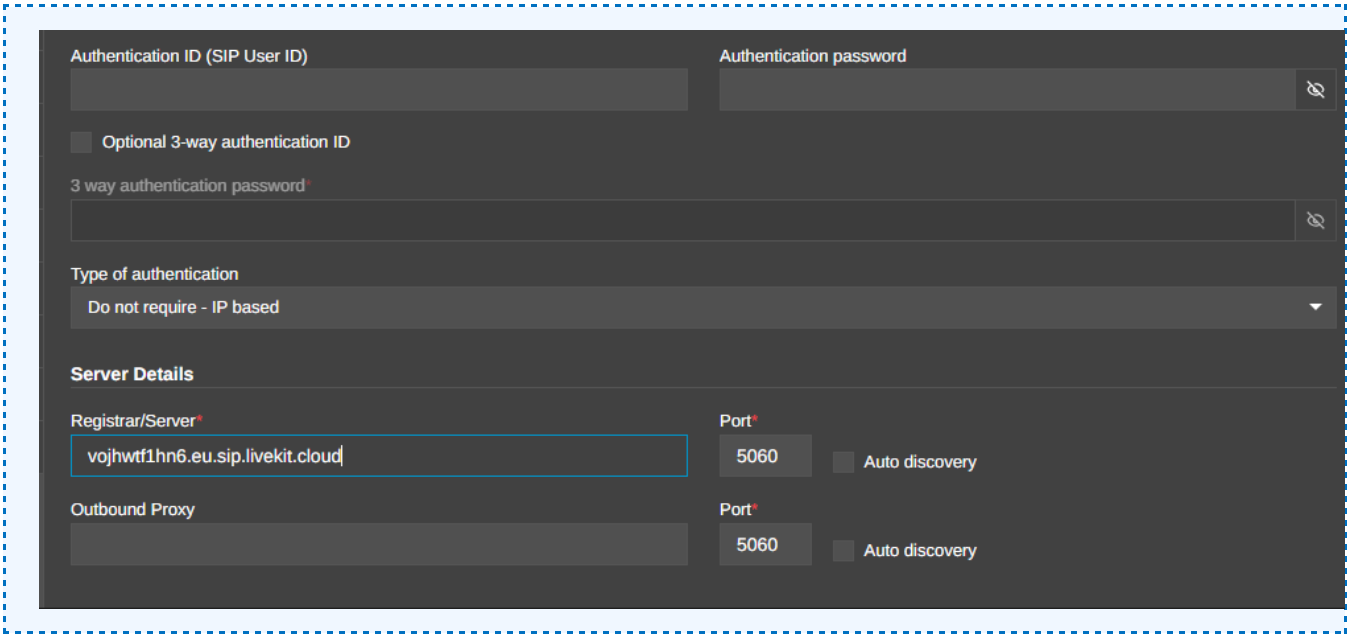
Field	What to Enter
Registrar / Server	`vojhwtf1hn6.eu.sip.livekit.cloud`
Port	5060 (this is the default; enter it explicitly)
Outbound Proxy	Leave this field completely blank
Type of Authentication	Do not require - IP Based
Authentication ID / SIP User ID	Leave completely blank

Authentication Password	Leave completely blank

⚠ The Authentication dropdown is critical - it must say 'Do not require - IP Based'

In some 3CX builds, the Authentication dropdown defaults to 'Register / Account Based'. If you leave it on this setting, 3CX will try to log in to Fonio's servers with a username and password. Fonio does not accept SIP logins from PBXes. The trunk will show connection errors and no calls will work.

Find the Authentication field and change it to: **Do not require - IP Based**. Then leave the Username and Password fields completely empty.



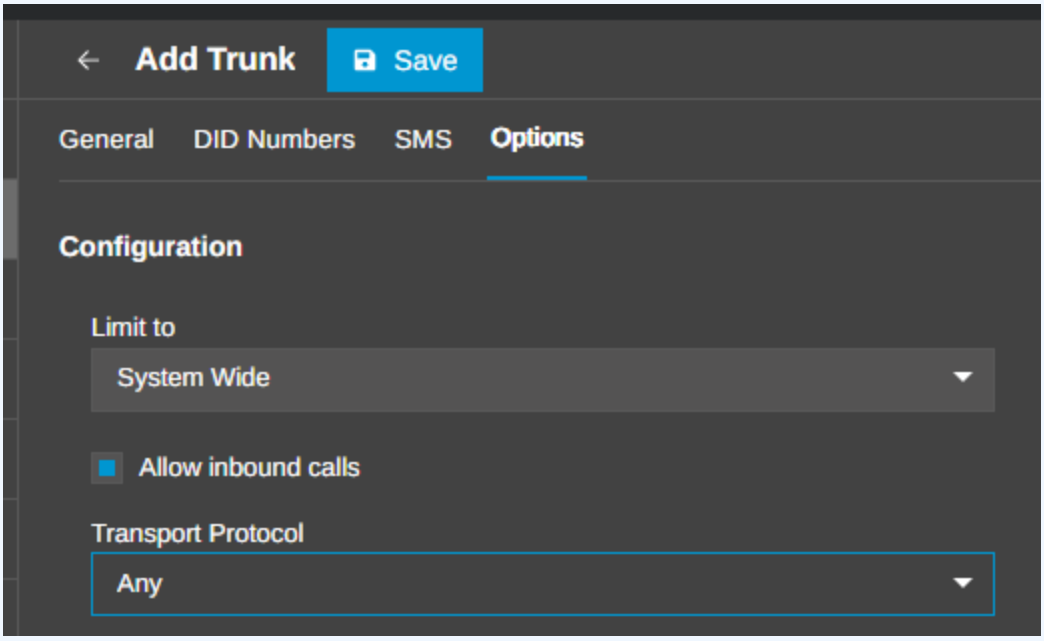
Step 17 — Navigate to the Options tab and set Transport to TCP

This is the most important single setting in the entire guide. Click the **Options tab** (the last tab, next to the **SMS tab**).

⚠ The Options tab is separate from the General tab - you must click it

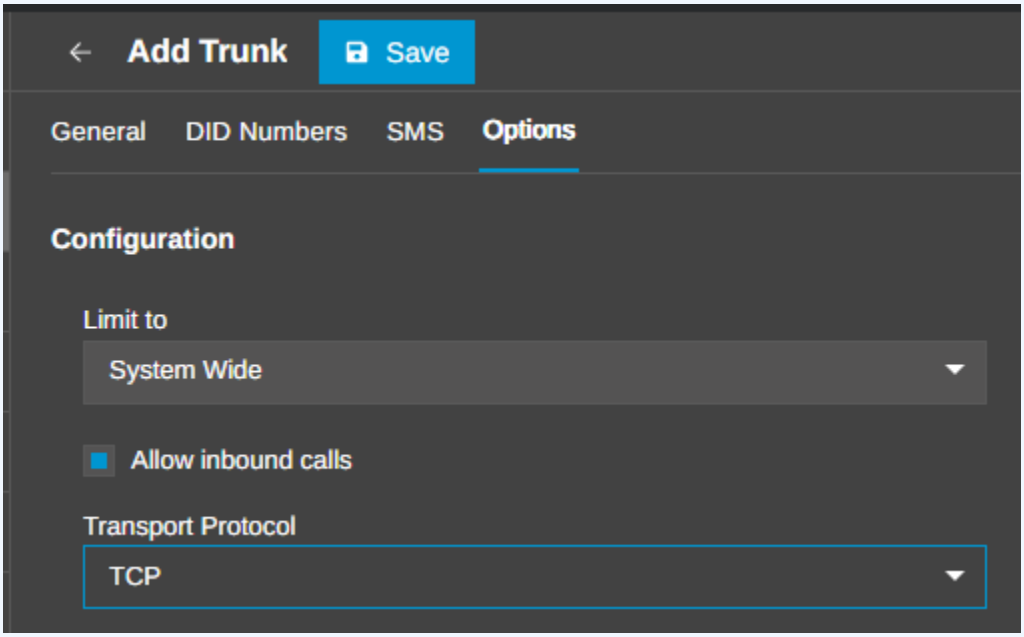
Many users fill in the General tab and click Save without ever opening the Options tab. The transport setting is NOT on the General tab. It is on the Options tab. If you save without changing it, the trunk will use UDP (the default), which causes calls to fail with Fonio's LiveKit servers.

Look for a row of tabs at the top of the trunk configuration page. Click the one labelled Options.



On the Options tab, find the **Transport Protocol** or **SIP Transport** dropdown. Change it from the default (UDP or Any) to **TCP**.

Setting	Value
Transport Protocol / SIP Transport	TCP ← Change this. Default is UDP. Must be TCP.



Step 18 Enable Caller ID Passthrough on the Options tab

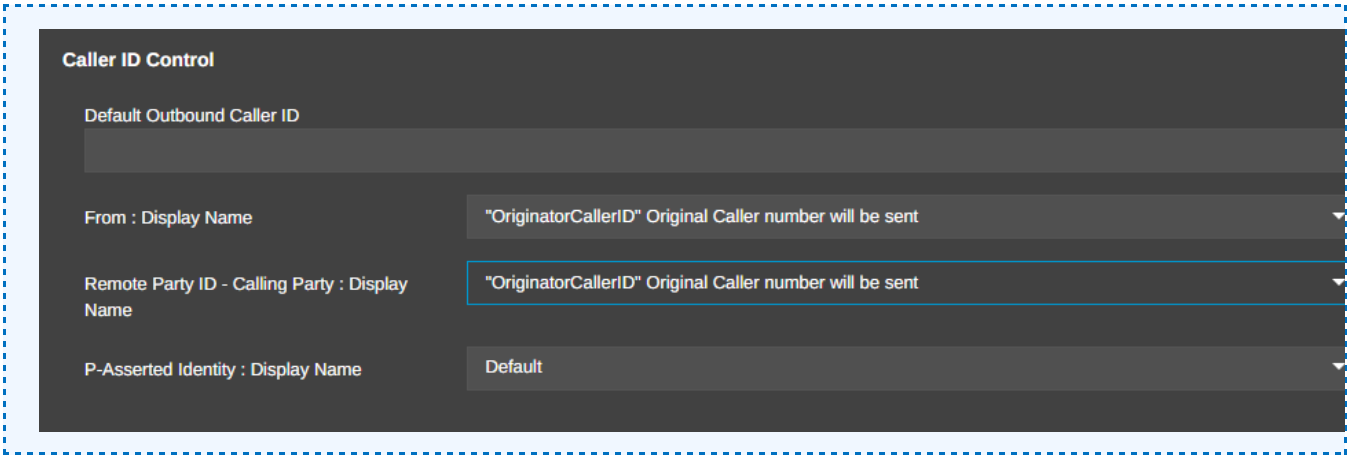
Still on the Options tab, find these two fields and configure them:

Field	Value
From: Display-Name	`OriginatorCallerID`
Remote-Party-ID	`OriginatorCallerID`

ℹ️ Why this matters - and why it is NOT optional

Without these two settings, Fonio sees every call as coming from your 3CX system's main number, not from the actual person calling. All calls look identical to Fonio. Your AI assistant cannot personalise the greeting, cannot identify returning callers, and your call logs will show the wrong number on every entry.

These settings tell 3CX to pass the real caller's original phone number through to Fonio unchanged. They must both be set, setting only one is not enough.



Step 19 Enable SIP REFER for AI call transfers

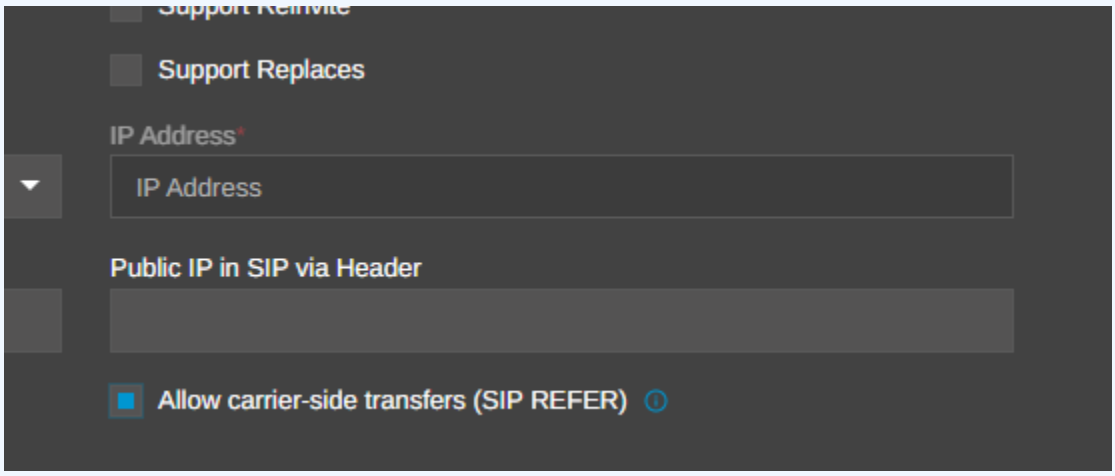
Still on the Options tab, find and enable this checkbox:

Setting	Value
Allow carrier-side transfers (SIP REFER)	Checked / Enabled, tick this box

ℹ️ What SIP REFER does and why you need it

SIP REFER is the technical mechanism that allows Fonio's AI to transfer a caller to a human. When the AI decides to hand off a call (for example, the caller asks to speak to a person), Fonio sends a SIP REFER message to 3CX saying 'please transfer this call to extension 100'.

If this setting is not enabled, Fonio will try to transfer the call but 3CX will reject the request. The caller will hear silence or the call will drop. Every customer who wants the AI to hand off to humans needs this enabled.



Step 20 **Save the trunk**

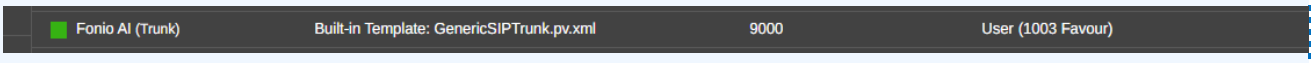
Click OK or Save to save the Fonio trunk. The trunk will appear in your Voice & Chat trunks list.

✅ **What a correctly saved Fonio trunk looks like**

Name: Fonio AI (or whatever you named it)

Main number: 9000 (or your SIP number)

If the trunk shows a red error indicator, see the Troubleshooting section at the end of this guide.



Task 4 - Create the Routing Rule

The Outbound Rule tells 3CX how to send calls TO Fonio.

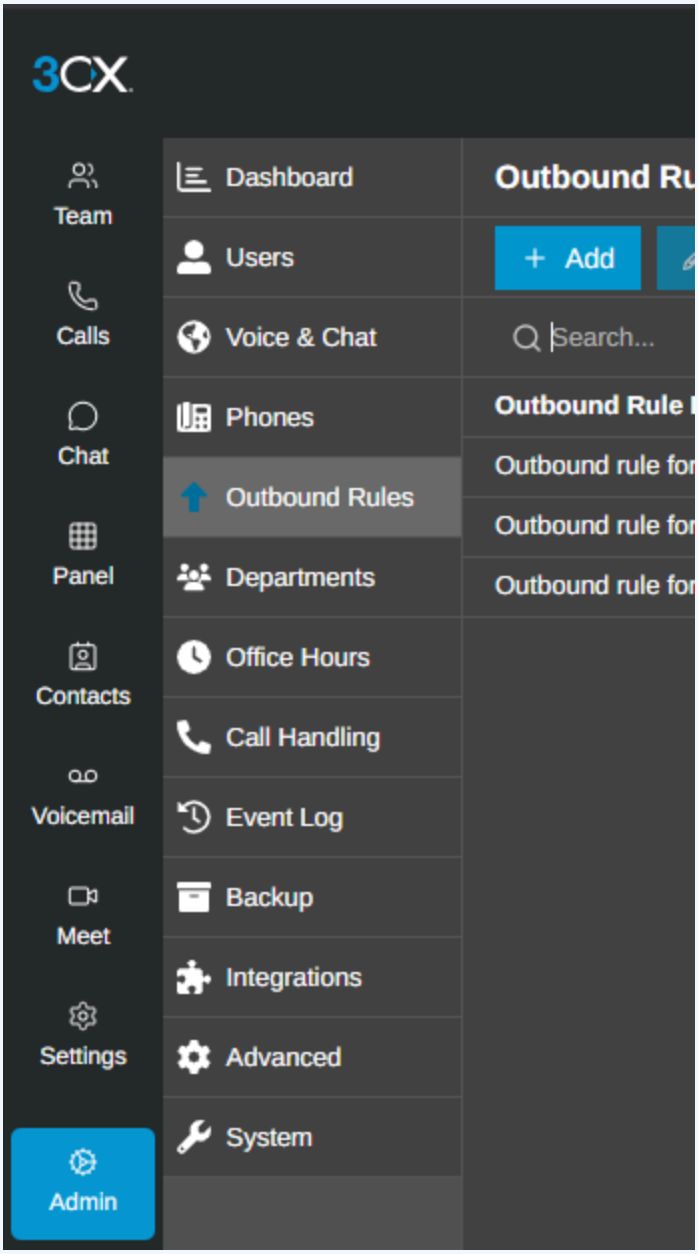
Outbound Rule

The Outbound Rule is the bridge. When 3CX receives a call that should go to Fonio, it needs to know which trunk to use and how to format the destination number. The Outbound Rule defines this.

> ℹ️ **Why do I need an Outbound Rule for what looks like an inbound use case?**
>
> It seems counterintuitive. Fonio is handling INCOMING calls, so why do you need an OUTBOUND rule?
>
> Here is the answer: when 3CX wants to send a call to Fonio, it looks at its Outbound Rules to decide which trunk to use. Even though the end result is Fonio answering an inbound call, from 3CX's perspective it is making an outbound call to Fonio's SIP number. The Outbound Rule says: 'when a call is made to a number starting with 999, strip those digits and send it out through the Fonio trunk.'

Step 21 — Navigate to Outbound Rules

In the 3CX Admin Console: **Admin Cog** → **Outbound Rules**. You will see any existing rules. Click **Add Rule**.



Step 22 **Fill in the Outbound Rule**

Outbound Rule Field	Value to Enter
Rule Name	Route to Fonio
Calls to numbers starting with prefix	999
Calls from extensions	Leave blank - this rule applies to ALL extensions

Calls to numbers with a length of	Leave blank
Route 1 - Make calls on	Fonio AI (select the Fonio trunk you created in Task 3)
Route 1 - Strip Digits	3 (this removes the 3-digit prefix '999' before the number is sent to Fonio)
Route 1 - Prepend	Leave blank
Routes 2 and 3	Leave blank - no fallback route needed

📋 How the outbound rule works - a concrete example

You (or your carrier rule) dial: `9999000`

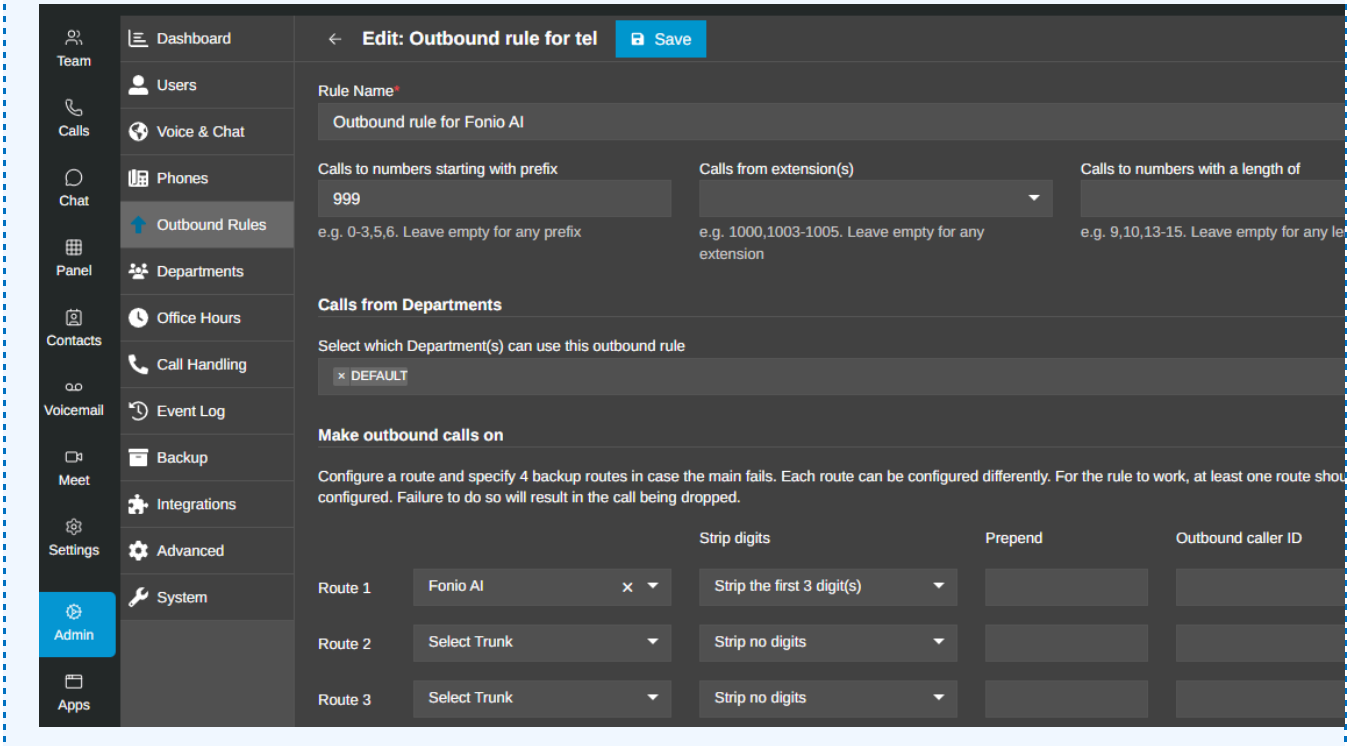
3CX sees: A number starting with prefix 999 → the 'Route to Fonio' rule matches

Strip 3 digits: 999 is removed from the start, leaving `9000`

3CX sends SIP INVITE to: `vojhwtf1hn6.eu.sip.livekit.cloud:5060` with destination number `9000` via TCP

Fonio receives: An inbound call for SIP number 9000 → routes to your assigned assistant

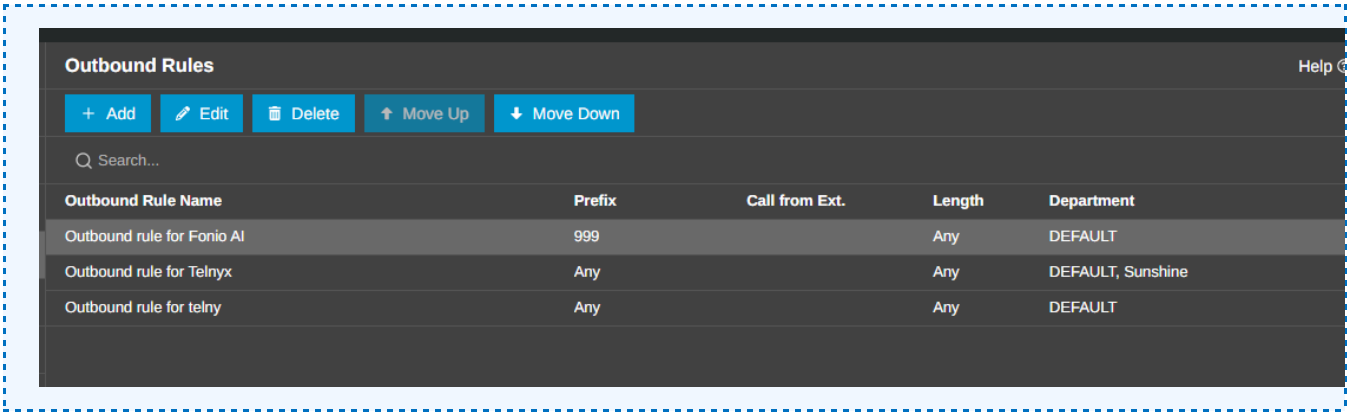
Result: Your Fonio AI assistant answers the call



Step 23 **Save the rule and move it to the TOP of the list**

Click Save. Then find the 'Route to Fonio' rule in the Outbound Rules list. It must be positioned ABOVE all other outbound rules. 3CX reads outbound rules from top to bottom and uses the first match. If another rule appears above yours and matches the 999 prefix (or has no prefix and catches everything), your Fonio rule will never be reached.

Use the **drag handle or arrow buttons** on the left side of each rule to reorder them. Move 'Route to Fonio' to **position 1** at the very top.



✅ Phase 2 - 3CX Configuration is Complete

You have now completed all four tasks in 3CX:

- Network / External IP confirmed
- Fonio IP ranges whitelisted (143.223.88.0/21 and 161.115.160.0/19)
- Fonio SIP Trunk created (Name: Fonio AI, Transport: TCP, Auth: IP Based, OriginatorCallerID set, SIP REFER enabled)
- Outbound Rule created (prefix 999, strip 3, routes to Fonio trunk - positioned at top of list)

PHASE 3

Testing - Verify Everything Works

Four tests, in order. Do not skip any. Each one checks a different part of the connection.

Never assume the setup is working just because no errors appeared during configuration. Run both tests below in order. Each test checks a specific part of the connection. If one fails, you will know exactly which part to fix.

Test 1 - Internal Dial (Does the SIP trunk to Fonio work?)

This test confirms that the trunk between 3CX and Fonio is working. It does not involve your carrier or public number at all.

Step 24 **Dial Fonio from a 3CX extension**

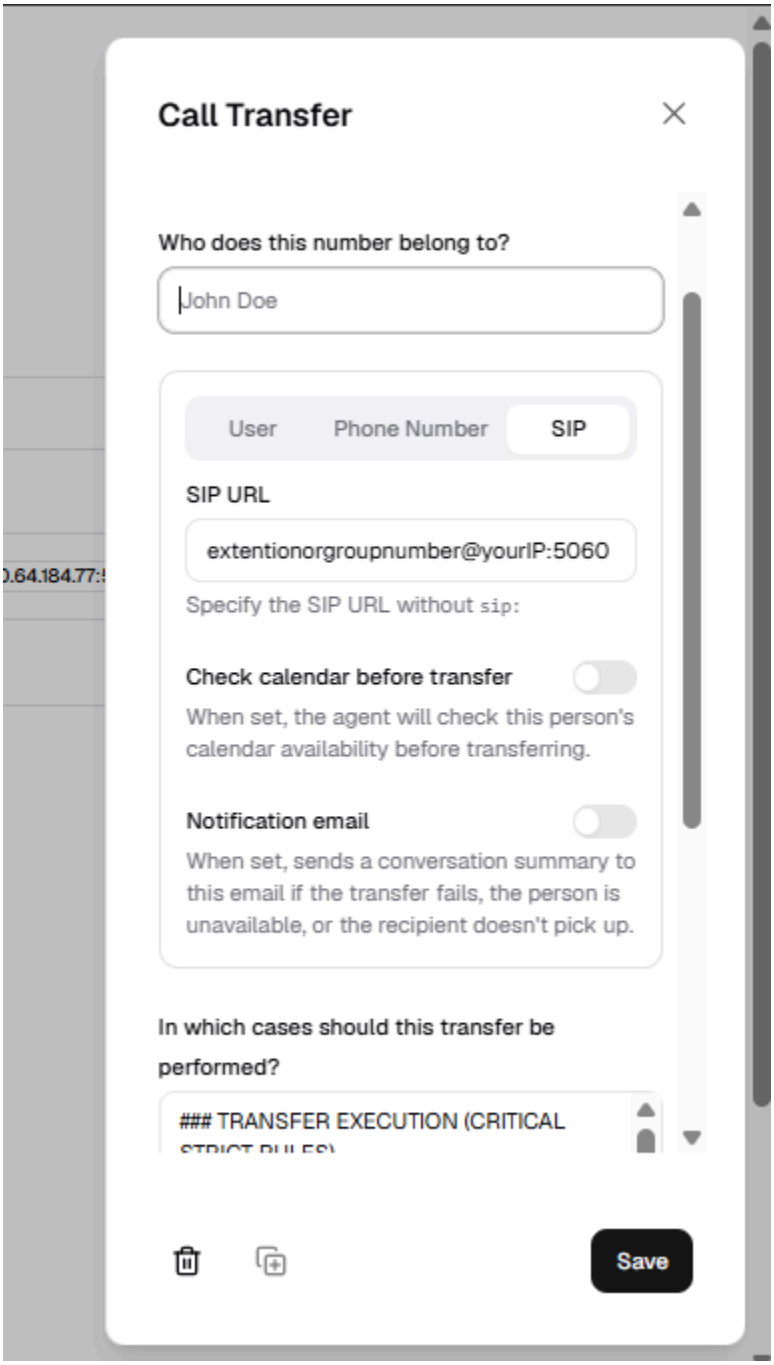
From any registered 3CX extension (web client, desktop app, mobile app, or IP phone), dial: `9999000`

(Replace 9000 with your actual Fonio SIP number if different.)

What you should experience	What it means
Phone connects within 3–5 seconds and your Fonio AI assistant speaks its start message	SUCCESS - the SIP trunk is working correctly
Fast busy tone immediately	FAIL - outbound rule is not matching. See Troubleshooting.
Rings but no answer / timeout after 30 seconds	FAIL - 3CX is sending the INVITE but Fonio is not responding. See Troubleshooting.
AI answers but you hear complete silence - no voice	FAIL - audio (RTP) is blocked. See Troubleshooting.

Test 2 - AI Transfer to a Human

This test confirms that Fonio can successfully transfer a caller back to a 3CX extension or group. This requires the SIP REFER setting (Step 19) and the Fonio Transfer tool configured.



Fonio Transfer tool configured inside the assistant settings, set the transfer target extension before testing.

Step 25 — Trigger a transfer during a live call

Make a call to Fonio (as in test 1). While speaking with the AI assistant, ask it to transfer you to a human or department, say whatever trigger phrase you have configured in the Fonio assistant's transfer settings (e.g. 'Can I speak to someone?').

Expected result	What it means
A pause of 1–3 seconds, then the target 3CX extension rings and a human answers	SUCCESS - SIP REFER transfer is working end-to-end
AI says it will transfer but call drops to silence or disconnects	SIP REFER is not enabled on the trunk (Step 19)
AI does not offer or acknowledge a transfer at all	The Fonio assistant has no transfer configuration. Set up transfer targets inside the Fonio assistant settings.

✅ **All 2 tests passing - your setup is complete**

If all two tests pass, your Fonio × 3CX integration is fully operational. Your AI assistant is answering calls, and human handoffs are working.

You are now ready to go live. Any calls to your public number will be answered by Fonio's AI with the ability to transfer to your team.

PHASE 4

Routing Calls to Fonio Automatically

How to forward calls from a Ring Group or User extension to the Fonio trunk

The tests in Phase 3 confirmed that dialling 999 + your Fonio SIP number from a 3CX extension reaches the Fonio AI. Phase 4 covers the next step: making that routing happen automatically, without anyone needing to manually dial a prefix. This is how real deployments work.

There are two scenarios covered in this phase:

Scenario A - Ring Group: When nobody in a ring group answers, the call overflows to Fonio automatically.

Scenario B - User Extension: When a user is unavailable (Away, Busy, or Do Not Disturb), their calls forward to Fonio automatically.

ℹ️ **Why this works - how the prefix travels through forwarding rules**

Both scenarios work using the same outbound rule you created in Phase 2 (Task 4). When 3CX forwards a call to an external number, it still processes that number through the outbound rules list.

This means if you set the forwarding destination to 9999000 (your prefix + Fonio SIP number), 3CX will match the 999 prefix, strip it, and route the call out through the Fonio trunk - exactly as if a person had dialled it manually.

The caller experience is seamless: the call is answered by Fonio’s AI with no manual intervention required.

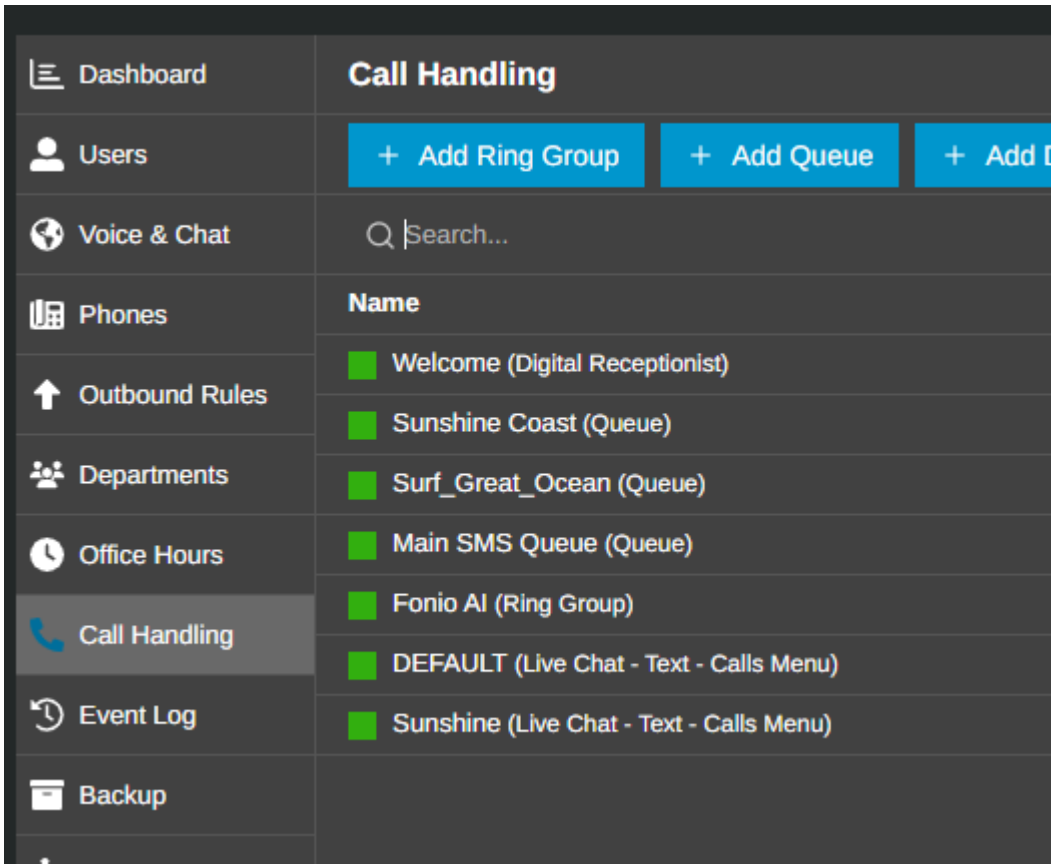
Scenario A - Forwarding from a Ring Group

This scenario is for businesses that want unanswered ring group calls to be handled by Fonio's AI instead of going to voicemail. When nobody in the group picks up within the configured timeout, the call automatically forwards to Fonio.

> ⚠️ **Before you start - enable external forwarding at system level**
>
> 3CX has a global setting that controls whether ring groups are allowed to forward calls to external numbers. If this setting is off, the forwarding will not work regardless of what you configure on the ring group itself.
>
> Go to: Admin Cog → Settings → General
>
> Find the option labelled: Allow external call forwarding (or Allow forwarding to external numbers in Ring Groups and Queues)
>
> Make sure this is enabled. If it was already on, you can skip this and proceed to Step 26.

Step 26 Navigate to Ring Groups

In the 3CX Admin Console, go to: Admin Cog → Ring Groups. You will see a list of all your configured ring groups. Open the ring group that should forward unanswered calls to Fonio.

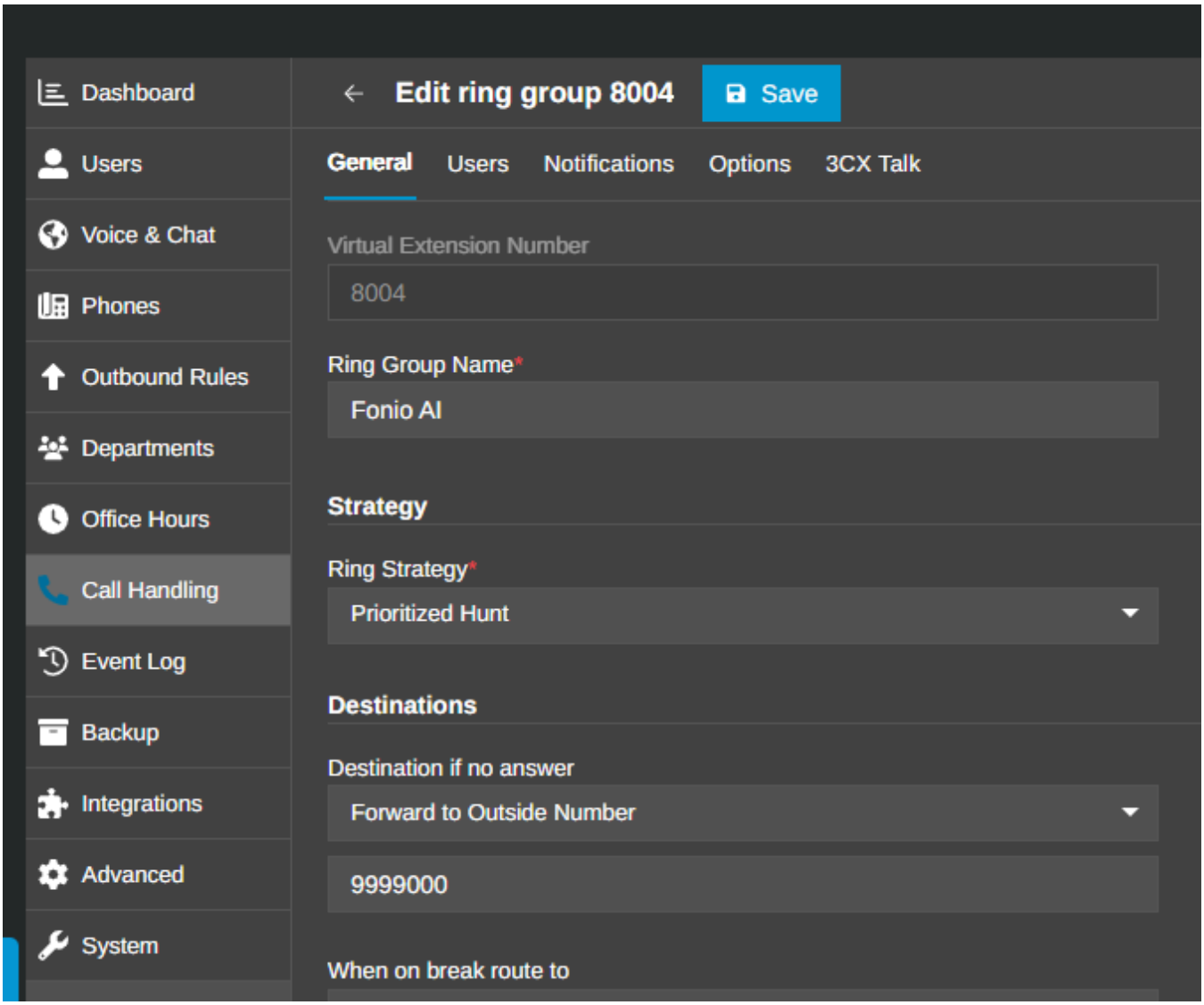


Step 27 Set the Destination if No Answer to Forward to Outside Number

Inside the ring group settings, look for the section titled Destination if No Answer (this appears after the ring group members list). Click the dropdown and select Forward to Outside Number.

In the number field that appears, enter your Fonio dial string:

9999000 (replace 9000 with your actual Fonio SIP number)



> ⚠️ **Important - the timeout setting**
>
> The ring group will only forward to Fonio after the configured ring timeout expires with no answer. Make sure your ring timeout (usually set in seconds on the same page) is set to a reasonable value - 20 to 30 seconds is standard.
>
> If the timeout is too short (e.g. 5 seconds), callers will be sent to Fonio before agents have a real chance to answer. If too long, callers wait unnecessarily.

Step 28 — Save the Ring Group

Click OK or Save to save the ring group configuration. The change takes effect immediately.

Step 29 — Test the Ring Group Forwarding

Call the ring group from a 3CX extension. Let it ring without answering until the timeout expires. Do not pick up on any extension in the group.

Expected result

After the timeout: the call stops ringing on the ring group members and Fonio's AI assistant answers automatically.

If the call drops to silence or ends: check that external forwarding is enabled globally (the warning step before Step 26) and that the number entered is exactly 999 + your Fonio SIP number with no spaces or symbols.

If the call goes to voicemail instead: the ring group has a separate voicemail destination configured that is overriding the Forward to Outside Number setting. Remove or clear the voicemail destination on the ring group.

Scenario B - Forwarding from a User Extension

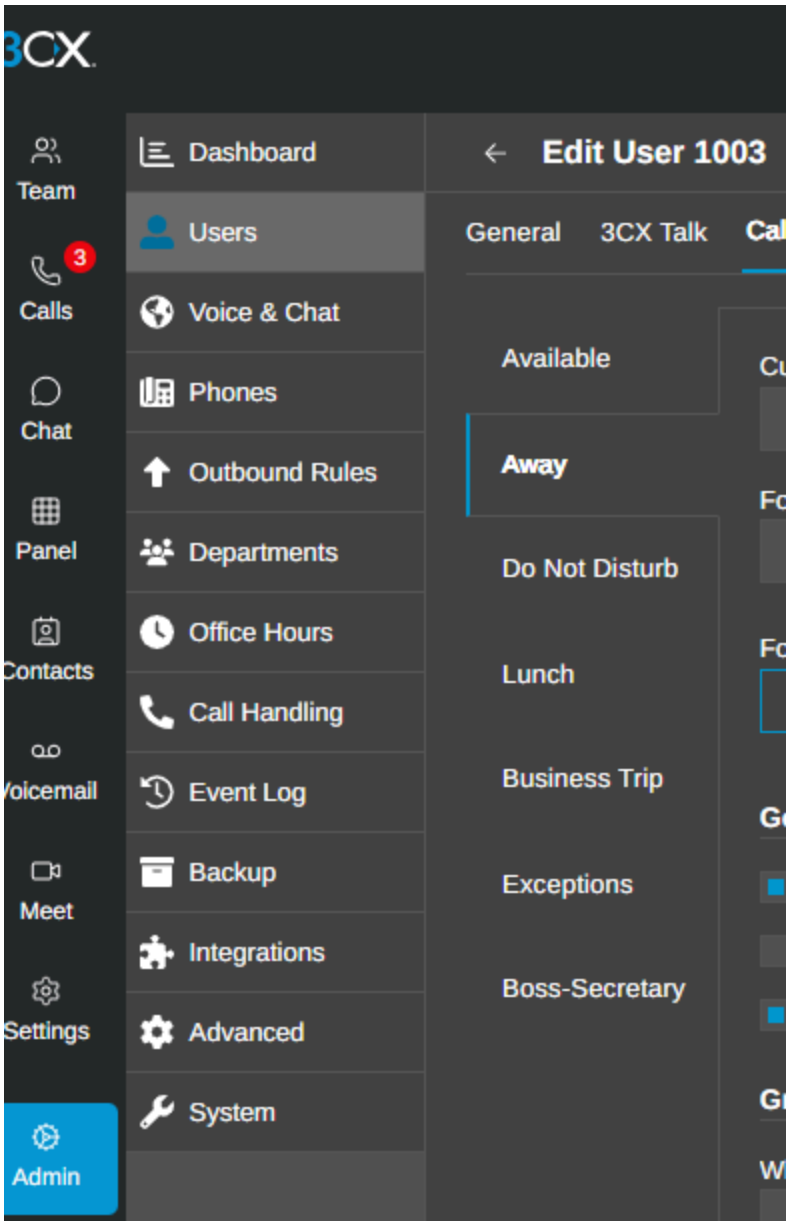
This scenario is for individual users who want their calls to go to Fonio's AI when they are unavailable. The forwarding is controlled by the user's status and their personal call forwarding rules in 3CX.

Step 30 Open the User's Forwarding Rules

There are two ways to access forwarding rules:

Option 1 - Admin sets it: In the 3CX Admin Console, go to Admin Cog → Users → open the user's profile → Forwarding Rules tab.

Option 2 - User sets it themselves: The user logs into their 3CX web client, goes to Settings → Call Forwarding.



Step 31 **Set the Forwarding Destination for the Desired Status**

You will see a list of status conditions: Available, Away, Do Not Disturb (DND), and Out of Office (if configured). Each status has its own forwarding rule.

For the status you want to forward to Fonio, change the destination to:

External Number → 9999000 (replace 9000 with your actual Fonio SIP number)

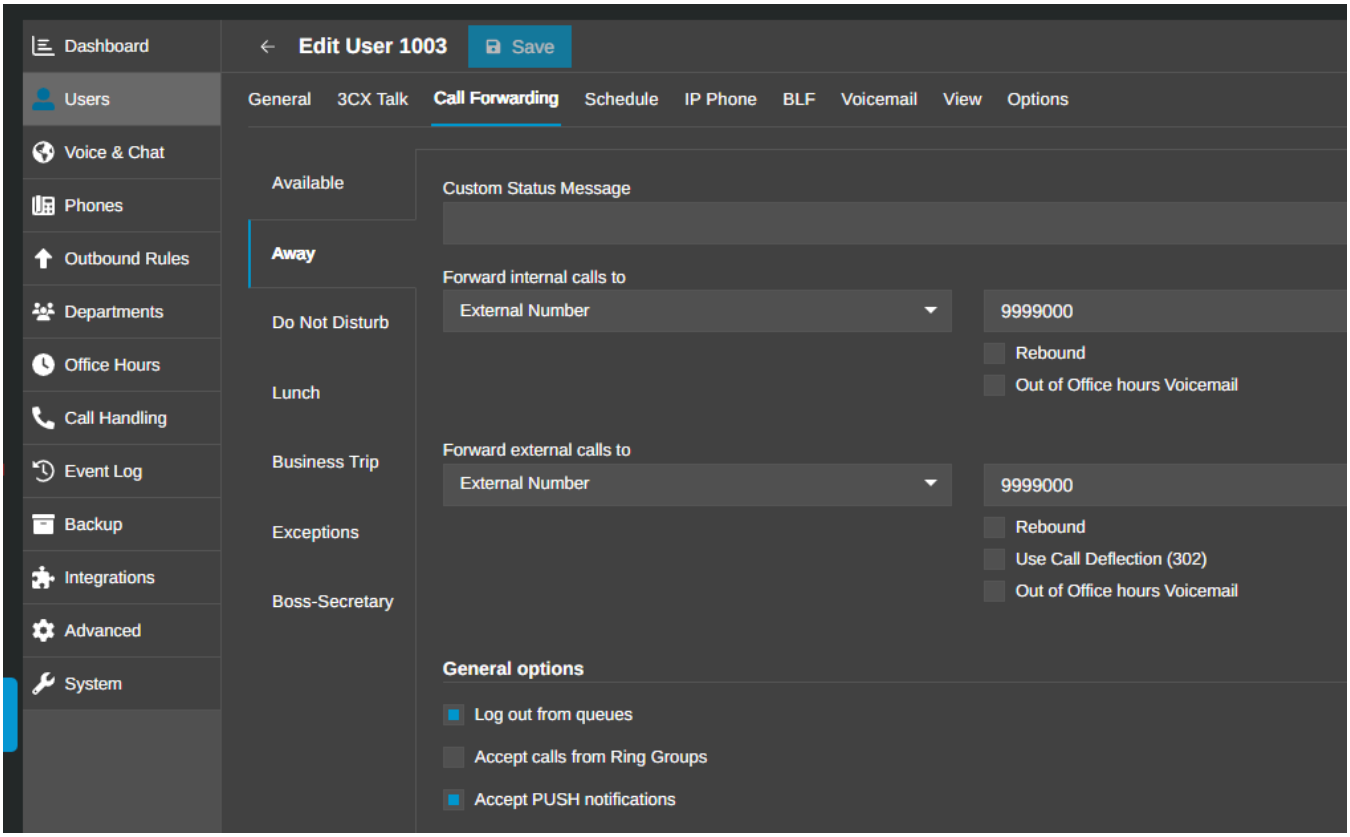
ℹ️ **Which status to use - guidance by use case**

Away: Use this when the user is temporarily away from their desk. They can manually set themselves to Away in the 3CX client to activate Fonio.

Do Not Disturb (DND): Use this when the user should not be interrupted at all. All calls go directly to Fonio without ringing the extension first.

Out of Office: Use this for planned absences (holidays, after-hours). 3CX can automatically activate this status based on office hours.

You can configure multiple statuses to all forward to Fonio if needed.



Step 32 Save and Activate the Status

Click Save to save the forwarding rule. To test it, the user (or admin) then changes their status to the one you just configured (e.g. Away).

In the 3CX web client or app, the user clicks on their status indicator (usually shown as a coloured dot next to their name) and selects Away.

[Screenshot: 3CX web client - user changing their status to Away]

Step 33 Test the User Extension Forwarding

From a separate phone or extension, call the user's extension number directly (not the ring group — dial their individual extension, e.g. 101).

Expected result

The extension should not ring at all (or ring very briefly). Fonio's AI assistant should answer the call automatically.

If the extension rings and is not forwarded: confirm the user's status is actually set to Away (not Available) in the 3CX client. The forwarding rule only activates when the user is in the correct status.

If the call ends without Fonio answering: check the number entered in the forwarding rule is exactly 999 + your Fonio SIP number and that the outbound rule from Phase 2 is still positioned first in the outbound rules list.

✅ **Phase 4 Complete - Automatic Call Routing is Active**

Ring Group forwarding: Unanswered ring group calls now route to Fonio automatically after the timeout.

User extension forwarding: Individual users can activate Fonio call handling by switching their status.

Your Fonio × 3CX integration is now fully configured for both manual and automatic call routing. Callers are handled by Fonio's AI whether they are forwarded from a ring group, an unavailable extension, or dialled directly via the Fonio prefix.

Troubleshooting - Solutions to Every Common Problem

Use this section if either tests failed, or if you experience issues after going live. Find the symptom that matches your situation.

Symptom	Most Likely Cause	Fix
Cannot find 'Generic SIP Trunk (IP Based)' in the provider list	Wrong Country selected, or 3CX license does not support it	**Fix:** Set Country to 'Worldwide' (not your actual country). If still not visible, your 3CX plan (Free/SMB) does not support this trunk type - upgrade to Business/PRO or self-host.
Fonio trunk shows RED or Error status (not Idle)	Authentication set to credential-based and 3CX is trying to register	**Fix:** Open the Fonio trunk → General tab → set Authentication to 'Do not require - IP Based'. Clear username and password fields. Save. The trunk should return to Idle.
Dialling 9999000 gives fast busy / No route available	Outbound rule not matching or in wrong position	**Fix:** Check: prefix is exactly '999', strip digits = 3, Route 1 = Fonio AI trunk. Confirm the rule is positioned FIRST in the outbound rules list above any catch-all rule.

Call connects but both sides hear complete silence	UDP blocked - RTP audio cannot travel between 3CX and Fonio	**Fix:** Open UDP ports 10000–20000 in both directions at your server/cloud firewall for Fonio's IP ranges. SIP signaling (TCP 5060) being open is not enough - audio needs separate UDP ports.
One-way audio — caller can hear AI but AI cannot hear caller (or vice versa)	3CX external IP is wrong - announcing a private address in SDP	**Fix:** Admin Cog → Advanced → Network → confirm External IP is your real public IP. If dynamic, enable STUN. Check whatismyip.com and match what 3CX shows.
AI answers but shows wrong caller number in Fonio call history	OriginatorCallerID not set on trunk	**Fix:** Fonio trunk → Options tab → set From: Display-Name = OriginatorCallerID AND Remote-Party-ID = OriginatorCallerID. Both must be set.
AI transfer drops the call / caller hears silence after transfer attempt	SIP REFER not enabled	**Fix:** Fonio trunk → Options tab → 'Allow carrier-side transfers (SIP REFER)' is ticked.
Fonio says SIP number is already taken	Another Fonio customer claimed that SIP number	**Fix:** Choose a different number (e.g. 8500, 7251). Update the Fonio trunk's Main Trunk Number.
Fonio answers but AI says nothing / silence on answer	No start message configured on the assistant	**Fix:** Go to Fonio → your assistant → Start Message. Add a message the AI should speak when it answers. Save.
Fonio answers with wrong assistant	Multiple assistants and SIP number assigned to wrong one	**Fix:** Go to Fonio → Phone Numbers → confirm that your SIP number (9000) is assigned to the correct assistant.

📧 **Still stuck after checking everything above?**

support@fonio.ai with the following information:

- Your 3CX version (V18 or V20) and deployment type (Self-hosted / Dedicated Hosted)
- Your Fonio SIP number
- The exact symptom and which test step it failed on
- A screenshot of your Fonio trunk Options tab in 3CX
- A screenshot of your Fonio trunk General tab in 3CX

The more detail you provide, the faster support can help you.

Quick Reference - All Key Values at a Glance

Print or bookmark this page. These are all the values you need for the connection.

Fonio Side

Parameter	Value
Fonio SIP Server (Registrar)	`vojhwtf1hn6.eu.sip.livekit.cloud`
SIP Port	`5060`
Transport Protocol	TCP - mandatory, not UDP
Authentication	None - IP-based only, no username/password
Fonio IP Range 1	`143.223.88.0/21`
Fonio IP Range 2	`161.115.160.0/19`
Voice Audio (RTP) Ports	UDP 10000–20000 - must stay open even though SIP is TCP
Your Fonio SIP Number	(your choice — e.g. 9000)

3CX Side

3CX Setting	Value
Trunk Type to Select	Country: Generic → Provider: Generic SIP Trunk (IP Based)
Main Trunk Number	Same as your Fonio SIP number - identical format, no + sign
Authentication Setting	Do not require - IP Based
Username and Password	Both blank
Registrar field	`vojhwtf1hn6.eu.sip.livekit.cloud : 5060`
Outbound Proxy	Blank
Transport (Options tab)	TCP - CRITICAL
From: Display-Name (Options tab)	`OriginatorCallerID`

Remote-Party-ID (Options tab)	`OriginatorCallerID`
SIP REFER (Options tab)	Enabled / Checked
Expected trunk status after save	Idle - this is correct, NOT an error

Routing Rules

Rule	Configuration
Outbound Rule - Name	Route to Fonio
Outbound Rule - Prefix	`999`
Outbound Rule - Strip Digits	`3`
Outbound Rule - Route via	Fonio AI trunk
Outbound Rule - Position	FIRST in the outbound rules list
To dial Fonio from a 3CX extension	`999 + your SIP number  (e.g. 9999000)`

Document prepared by: Favour Igbinevbo (therhymzoai) - VoIP & SIP Engineer

Covers: 3CX V20 / V18 + Fonio AI SIP Trunk integration

Support: support@fonio.ai