NeuraPBX

User Manual

Version 1.0.1 · August 2026
Self-hosted AI-native IP PBX · neurapbx.com

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NeuraPBX is an AI-first enterprise IP PBX built on FreeSWITCH, with a Node.js REST API, a React admin dashboard, a Python AI service (Whisper / faster-whisper transcription + OpenAI or on-prem Ollama language models), native Windows & macOS softphones, and full traditional PBX functionality.

This manual is a detailed, feature-by-feature guide. Every feature has a short description, a **Set up** procedure (what to click in the dashboard) and a **Use it** procedure (how it behaves for callers, agents, and admins).

How to use this manual

Each entry covers one feature. Work top-to-bottom the first time you deploy, or jump to a specific feature as needed. To follow the call-handling steps you'll want **two extensions registered on two softphones** (e.g. 100 and 101) so you can place real calls between them.

Before you begin

- **Sign in** to the dashboard at your server address (e.g. `http://<server-ip>:8080` — the port is `WEB_PORT` in `.env`, 8080 by default) with your admin account.
- **Register at least two SIP softphones** (Zoiper, Linphone, MicroSIP, or the built-in Web Phone) to two extensions so you can place calls while learning.
- **Confirm the stack is up** — `docker compose ps` should show FreeSWITCH, the API, the web app, and the AI service all "Up".
- Most changes (extensions, IVRs, ring groups, queues, conferences, time conditions, trunks, routes, feature codes, Follow Me…) **auto-regenerate and reload the FreeSWITCH dialplan** — no restart needed. If a change doesn't take effect, let any in-progress call end first.

Contents

Getting started

Signing in

Open the dashboard URL and sign in. Options:

- **Email + password** — the admin account from `.env` on first run; add more users on the Extensions / Users pages.
- **Two-factor (2FA)** — enable a TOTP authenticator from your profile; sign-in then asks for the 6-digit code (recovery codes are issued once).
- **Single sign-on (optional)** — if the admin configured it, a **Continue with Google / Microsoft / GitHub** button appears; **LDAP / Active Directory** logins work with directory credentials.

Roles: **superadmin** (everything), **admin** (all configuration), **supervisor** (monitoring), **user** (My Portal only).

Your plan

Entitlements come from a license key installed under **System → License** (which also shows this server's **Machine ID**, tier, and expiry). With no key you run the **Free tier** (5 extensions, core PBX, no trunk) plus a 30-day trial of 1 SIP trunk + full AI. Paid plans: **Starter / Business / Enterprise** subscriptions - each includes an AI bundle for every user at no extra per-seat cost - or **perpetual** offline packages (Micro→X-Large, machine-locked, no AI). Features unlock by tier — see **§22 Licensing**.

Part 1 — Core PBX Features

Everything below is configured from the web dashboard: extensions, call routing, queues, conferences, trunks, voicemail, recordings, monitoring, devices, the web phone, the self-service portal, the operator console, reporting, and system-wide features.

1. Extensions

Dashboard: Extensions page. Extensions are the individual phone lines (users/desks). Each has a number, name, password, and voicemail.

Set up 1. Open **Extensions → + New Extension**. 2. Enter **Extension Number** (e.g. 100), **Full Name**, **Email**, and **Password**. 3. Optionally set **Department**, **Context** (default `default`), **Max Calls** (1–10), and **Call Timeout** (5–120 s). 4. Leave **Voicemail** enabled; leave **Record Calls** off unless you need recording. 5. **Save**. Repeat for a second extension (e.g. 101).

Use it 1. Register a softphone to 100 with its password, and a second to 101. 2. Both should show **Registered**. 3. Dial 101 from 100 — the call connects with two-way audio. The **Email** field matters: it links the user to **My Portal** (§15) and voicemail-to-email (§10).

2. IVR Menus

Dashboard: IVR Menus page. An IVR plays a greeting and routes callers by the digit they press.

Set up 1. Open **IVR Menus → + New IVR Menu**. 2. Enter a **Menu Name** (e.g. `main_menu`) and description. 3. Enter the **Greeting** text (spoken by TTS), e.g. "Press 1 for Sales, 2 for Support." 4. Set **Timeout** (default 10 s) and **Max Failures / Max Timeouts** (default 3 each). 5. **+ Add Option** for each digit: **Digit**, **Action** (transfer / ivr / queue / voicemail / hangup), **Destination**, description. 6. **Save**.

Use it 1. Point an inbound route, time condition, or extension at the IVR's name/extension. 2. Call in — the greeting plays; each digit routes to its destination; timeouts/invalid entries follow the failure behavior. For a spoken menu instead of press-1, enable **Natural-Language IVR** (Part 2, §6).

3. Ring Groups

Dashboard: Ring Groups page. Rings several extensions at once (or in sequence) under one number.

Set up 1. Open **Ring Groups → + New Ring Group**. 2. Enter **Group Name**, **Extension** (the dial-in number, e.g. 6001), description. 3. Choose **Ring Strategy**: simultaneous, sequential, round_robin, or random. 4. Set **Timeout** (default 30 s). 5. **Add** each member extension (type the number, press Enter). 6. **Save**.

Use it - Dial the group extension: members ring per the strategy (all at once, or one at a time). Answering on one member stops the others. If no one answers within the timeout, the call follows the group's failover destination.

4. Call Queues (ACD)

Dashboard: Queues page. Holds callers in line and distributes them to agents by strategy.

Set up 1. Open **Queues → + New Queue**. 2. Enter **Name** and **Extension** (e.g. sales / 5000). 3. Choose **Strategy**: Round-Robin, Least-Recent, Fewest Calls, or Random. 4. Set **Max wait** (default 300 s); optionally enable **Announce queue position** with a frequency (default 30 s). 5. **Add Agents**: each agent's **Extension** and **Penalty** (lower rings first, default 0). 6. **Save**.

Use it - Dial the queue extension: the caller hears hold music and (if enabled) position announcements, then connects to an agent per the strategy. If all agents are busy, the caller waits until an agent frees up or **Max wait** elapses. Live queue stats appear on the **Reports** wallboard (§17).

5. Conference Bridges

Dashboard: Conferences page. Meet-Me rooms let multiple people join one call with a room number and PIN.

Set up 1. Open **Conferences → + New Room**. 2. Enter **Name**, **Extension** (e.g. 8000), description. 3. Set a **User PIN** and a **Moderator PIN**. 4. Set **Max members** (default 20); optionally **Record conferences**. 5. **Save**. (Change the default room 8000's PINs before use.)

Use it 1. Dial the room extension from two or more phones; enter the **User PIN** — everyone can hear each other. 2. To join as moderator, dial the room extension then `#` and the **Moderator PIN** (moderator controls, e.g. recording). 3. If recording was enabled, a file is saved to **Recordings** (§11) after the room empties.

6. Time Conditions & Holiday Routing

Dashboard: Time Conditions page. Routes calls differently by day/time and on public holidays.

Set up 1. **Time Groups → + Group**: name it, select active days, set Start/End (e.g. Mon–Fri 09:00–17:00). 2. **Public Holidays → + Add**: name and date. 3. **Routing Conditions → + Condition**: **Name**, **Node extension** (e.g. 6000), the **Time Group**, an **In-hours** destination, and an **After-hours/holiday** destination. 4. **Save**, then point an inbound route (or feature) at the condition's node extension.

Use it - In-hours calls to the node go to the day destination (e.g. the AI receptionist / IVR); after hours and on holidays they go to the night destination (e.g. night voicemail). Force night mode any time with the **Day/Night override** on PBX Features, or dial `*280`.

7. SIP Trunks

Dashboard: Trunks page. A trunk connects you to a SIP provider for outside calls.

Set up 1. Open **Trunks → + New Trunk**. 2. Enter **Name**, **Provider**, **Host**, **Port** (default 5060), **Username**, **Password**. 3. Choose **Transport** (UDP/TCP/TLS) and confirm **Codecs** (default PCMU, PCMA, G722). 4. **Save** — the trunk should show **Active** in the list.

Use it - Place an outbound call that matches an outbound route (§8) — it connects through the trunk. Inbound calls to your provider DID arrive at the PBX (watch **Call Logs**, §17). Trunk count is limited by your plan (§22).

8. Outbound Routes (incl. PIN Sets & E911)

Dashboard: Outbound Routes page. Decide which trunk handles a dialed number, reformat the number, and optionally require a PIN or mark emergency calls.

Set up 1. Open **Outbound Routes → + New Route**. 2. Enter **Name** and **Pattern** (regex, e.g. `^(\d{10,11})$`). 3. Set **Strip digits** / **Prepend** if the trunk needs reformatting. 4. Enter **Trunk IDs** in failover order (comma-separated, e.g. `1,2`) and **Priority** (default 100). 5. Optionally attach a **PIN set** (from **PIN Sets**) to require a code before dialing (class of service). 6. For emergencies, create a separate route, tick **Emergency route**, leave the PIN set empty. 7. Optionally set **E911 locations** (per-extension street address + callback ELIN). **Test emergency calling before you rely on it**: ask your carrier how to place a safe test call (many US carriers answer `933` with the address they have on file) and confirm the address read back matches each location — never assume a route works because it saved.

Use it - Dial a number matching the pattern — it's offered to the correct trunk by priority (see the route used in Call Logs). PIN-protected routes prompt for the code; emergency numbers always route first and never prompt for a PIN.

9. Feature Codes & Call Parking

Dashboard: Feature Codes page. Star codes trigger features from the keypad; parking lots hold a call for pickup elsewhere.

Set up 1. Open **Feature Codes**. For each row (Voicemail Check, Call Forward on/off, Pickup, Echo Test, Supervisor Spy/Whisper/Barge...) edit the **Code** if you want a non-default, and toggle **Enabled**. 2. **Save** any code you changed. 3. Review the **Parking Lot** slot numbers/status.

Use it (defaults; see the Appendix) - `*97` voicemail check · `*72` +number / `*73` call-forward on/off · `*8` pickup · echo test code (self-test) · park a call (announces a slot) then dial the slot from any phone to retrieve it · supervisor spy/whisper/barge code + extension. Disabled codes no longer respond.

10. Voicemail

Dashboard: Voicemail page. Every extension can have a PIN-protected mailbox, reachable from the dashboard or a handset.

Set up 1. Open **Voicemail**, pick a box, enter a **PIN** (4–8 digits), **Set PIN**. 2. Greetings are recorded from the handset (see Use it).

Use it 1. From the extension, dial `*97` and enter the PIN to log in; use handset **option 5** to record busy/unavailable greetings. (`*98` is direct access.) 2. Callers who reach the mailbox leave a message; it appears on the handset **and** on the Voicemail page (play, download, delete in the browser). 3. Turn on **Voicemail → Email** (Part 2, §2) to get transcribed messages by email.

11. Call Recordings

Dashboard: Recordings page. Calls recorded via a per-extension "Record Calls" setting (or conference recording) are listed for playback, download, transcription, and deletion.

Set up - Enable **Record Calls** on an extension (§1) or **Record conferences** on a room (§5). No other setup — recordings appear automatically.

Use it 1. Place and end a call on a recording-enabled extension. 2. On **Recordings**, the entry appears with duration and size — **Download** to play it. 3. Click the **AI transcribe** button for a transcript + sentiment tag (one click). 4. **Delete** removes it from the list.

12. Supervisor Monitoring (Spy / Whisper / Barge)

Dashboard: Monitoring page. A supervisor silently listens to, coaches, or joins a live call.

Set up - Open **Monitoring** and enter **your extension** in the input; the live-call table auto-refreshes every 5 s.

Use it - Find a live call and click **Spy** (listen only), **Whisper** (coach the agent; caller can't hear), or **Barge** (join as a 3-way). Your phone rings, then you're in; switch modes mid-call with DTMF 1/2/3, or dial `*33` +extension from a phone. The **Parked Calls** panel lists any parked calls.

13. Phone Provisioning (Devices)

Dashboard: Devices page. Auto-provisions desk phones with their account and BLF keys. Each vendor gets its own config format — Yealink and Fanvil key-value files, Grandstream/Polycom/Cisco/Snom XML.

Any SIP phone works regardless. Auto-provisioning only saves you typing: every SIP 2.0 handset can be registered by entering the extension, password and server by hand, whether or not its make appears below.

Vendor	Config format	File the phone requests	Verified on hardware
Yealink	key-value	`<MAC>.cfg`	**Yes**
Fanvil	key-value	`<MAC>.cfg`	Not yet
Grandstream	XML (P-values)	`cfg<MAC>.xml`	Not yet
Polycom	XML	`<MAC>-phone.cfg`	Not yet
Cisco **SPA/MPP**	XML	`<MAC>.xml`	Not yet
Snom	XML	`<MAC>.htm`	Not yet
Anything else	manual-setup sheet	`<MAC>.cfg`	—

"Not yet" means the template follows the vendor's published format but has not been confirmed against a physical handset of that make. A wrong parameter name does not raise an error — the phone ignores the line and comes up unregistered — so treat those as untested until you have provisioned one.

Cisco: this covers the SPA/MPP series running SIP firmware. Enterprise Cisco phones running SCCP against CallManager are not provisionable here.

An unrecognised vendor is served a plain-text sheet of the settings to enter by hand, deliberately rather than a guess at some other vendor's syntax — a config the phone silently ignores looks like provisioning already worked.

Set up 1. Open **Devices** → **+ Add Device**. 2. Enter the phone's **MAC** (12 hex), **Vendor**, **Model**; assign an **Extension** and **Label**. 3. Optionally add **BLF/Speed keys**: key #, watched extension, label. 4. **Save**, then **copy URL** — this gives the exact filename that vendor's phone asks for. 5. Point the phone's auto-provision server at that URL (or at the base `/provision/` path) and reboot it.

Use it - The phone boots, fetches its account + key layout, and registers under the assigned extension. Configured BLF keys light up when the watched extension is busy. Bulk-import devices via CSV on the System page.

14. Web Phone (Browser Softphone)

Dashboard: Web Phone page. A built-in WebRTC/SIP.js softphone — no separate app.

Set up 1. Open **Web Phone**; confirm the **WSS server** (default `wss://<host>:7443`) and SIP domain. 2. Enter an **Extension** and **Password**, click **Register** — the status shows **registered**. 3. WebRTC needs `mod_verto`/WSS (port 7443) enabled on FreeSWITCH.

Use it - Dial from the on-screen pad and press call — two-way audio over WebRTC. Incoming calls to that extension ring (or auto-answer) in the browser; the hangup button ends the call cleanly.

15. My Portal (User Control Panel)

Dashboard: My Portal page (end-user login). Lets users manage their own voicemail, Follow Me, speed dials, and recent calls without admin access. The user's login email must match their extension's Email.

Set up - Ensure the user's extension has an **Email** (§1) matching their login.

Use it (as the end user) 1. Sign in and open **My Portal**. 2. **My Voicemails** — play, download, delete in the browser. 3. **DND** — one-click do-not-disturb (calls go to voicemail/busy). Toggle off to resume ringing. 4. **My Follow Me** — enable, set destinations (comma-separated extensions/numbers), strategy, and desk-ring time; optional press-1 confirm so mobile voicemail can't swallow the call. 5. **My Speed Dials** — add name/number/code; dial `*0`+code to use. 6. **My Recent Calls** — your call history.

16. Operator Panel

Dashboard: Operator Panel page. A receptionist console: live extension status (BLF grid) + active calls with manual transfer/hangup.

Set up — none beyond having extensions.

Use it - Each tile shows status (green = idle, red = on a call, gray = offline). For an active call, type a destination in **Transfer to** and click **Transfer**, or click **End** to hang up.

17. Reports & Call Logs (CDR)

Dashboard: Reports page and Call Logs page. Read-only reporting: call volume, answer-rate SLA, queue wallboard, and a searchable CDR.

Set up — none; both read call history automatically.

Use it - **Reports**: Total Calls, Answered, Missed, Answer Rate, and (if a queue is active) a live agent/queue wallboard. - **Call Logs**: filter by extension / direction / disposition; each row shows duration and a recording indicator with links to the recording and AI analysis.

18. PBX Features

Dashboard: PBX Features page. A set of smaller call-handling modules, each with a simple add/edit/delete form, plus a system Day/Night override.

Set up (add entries as needed) - **Follow Me** — extension, destinations, strategy, primary ring timeout, enabled (admin can set this for any extension). - **Phonebook & Speed Dials** — name, number, speed code, department. - **Paging & Intercom Groups** — name, extension, members, mode (page = one-way announce; intercom = two-way). `*54` +extension intercoms one phone. - **Caller-ID Blacklist** — number/pattern + reason to block callers. - **CID Routing** — CID pattern → destination + priority (send VIPs straight to their manager). - **DISA** — name, extension, PIN, context to dial in and get an internal dial tone. - **Announcements** — name, extension, recording, then a destination to continue to. - **Group Voicemail (VM Blast)** — one message copied to every member's mailbox. - **Wake-up Calls** — extension, time, recurring. - **Tenants** — name, SIP domain, max ext (scaffold; not production-isolated).

Use it - Dial a paging/intercom group to announce or open two-way audio. Blocked callers (added from the dashboard's Caller-ID Blacklist) get a 603 Decline. Dial a DISA extension from outside, enter the PIN, get an internal dial tone. The **Day/Night override** (or `*280`) instantly switches day/night routing system-wide; named call-flow toggles use `*281` … `*289`.

19. System Administration

Dashboard: System page. System-wide configuration and integrations.

Set up 1. Review each **settings card** (SIP/RTP/Media, Email/SMTP, Security, Toll-Fraud, Integrations, Backups) and **Save**. 2. **Call Flow Toggles** → **+ Flow**: name, index, extension, day/night destinations (each gets a `*28X` code). 3. **Backups**: **Backup now** for a manual backup; the schedule (default 3 a.m. daily) + retention show here; one-click DB download; restore is manual. 4. **API Tokens** → **+ Token**: name, scope (read / read_write), **Create** — copy the token (shown once); send it as the `X-API-Token` header. 5. **Webhooks** → **+ Hook**: name, public URL, events (`call.started` / `call.ended`), secret (payloads are HMAC-signed). 6. **CDR Billing** → **+ Rate**: name, pattern, per-minute rate, connect fee. 7. **Scheduled Bells** → **+ Bell**: name, time, page extension (school bells / shift changes). 8. **Bulk Import/Export**: export or paste CSV for phonebook / devices / extensions / blacklist. 9. **Audit Log**: every admin change (who, what, when) with secrets redacted.

Use it - Monitoring is exposed at `/metrics` (Prometheus) and `/health`. Tokens/webhooks authenticate and fire on the configured events; the audit log records all admin actions.

20. General Settings & Network

Dashboard: Settings page. A simpler, general-purpose settings form.

Set up 1. **General** — Company Name, Timezone, Language. 2. **Call Settings** — Max Extensions, Recording Format (WAV/MP3), Recording Retention (days). 3. **Network** — the Server IP/Hostname FreeSWITCH uses for signaling/audio (leave blank to keep current). 4. **AI Features** — quick toggles for Enable AI, Auto-transcribe all calls, Sentiment Analysis, Voicemail to Email. 5. **Save Settings**.

Use it - After a network address change, re-register softphones if needed. With **Auto-transcribe all calls** on, recordings are transcribed without clicking the AI button; with **Voicemail to Email** on, box owners get an email per message.

21. Roles & Permissions (RBAC)

Dashboard: System → Roles & Permissions. Custom roles with per-module read/write permissions, on top of the built-in superadmin/admin/supervisor/user roles.

Set up 1. **+ Role**: a **name** (lowercase, 2–30 chars) and description. 2. Check the desired boxes in the **module × read/write** matrix (e.g. `call-logs:read`, `recordings:read` for a reporting-only role). 3. **Save**, then assign the role to a user.

Use it - The user sees only the modules/actions granted; anything else is blocked with an authorization error. Built-in roles are protected and can't be modified/deleted.

22. Licensing

Dashboard: System → License. Shows the current tier, usage against limits (extensions, trunks), the AI minutes processed this month (a counter, not a quota), and lets you install/remove a key.

Set up 1. Open **System → License** and review the **tier badge**, **Machine ID**, expiry, and usage meters. 2. To activate, paste the key (`NPBX-…`) and **Install**. To return to Free, **Remove**.

Use it / tiers - **Free** (no key): 5 ext, core PBX, no trunk + a 30-day trial (1 trunk + AI). **Starter** 25 ext / 2 trunks · **Business** 100 ext / 8 trunks · **Enterprise** 500 ext / unlimited — annual. AI is **included** with every paid plan for every user: Starter gets **AI Essentials**, Business and Enterprise both get **AI Ultimate** (every AI feature). There is no per-seat AI charge and no AI usage metering. **Perpetual** (one-time, offline, machine-locked, no AI): Micro (8/2) → X-Large (256/32); add-ons scale to 512 ext / 64 trunks. - Features unlock by tier: queues, conferences, time conditions, feature codes need **Starter+**; provisioning, monitoring, My Portal, media, outbound routes need **Business+**; API tokens, webhooks, audit, multi-tenant, call flows need **Enterprise**. AI needs a plan that includes it (any paid subscription, or the 30-day trial); perpetual keys carry no AI. Invalid keys are rejected without changing the current license.

23. Media, Prompts & Fax

Dashboard: Media & Prompts page. Music on Hold, system recordings/prompts, and inbound fax-to-email.

Set up 1. **Music on Hold Classes → + Class**: name, Source (Files or a Stream URL), upload `.wav`/`.mp3`/`.ogg`. Assign the class to queues. 2. **System Recordings** → name + upload a prompt file → **Upload** (reference it from IVRs / Announcements). 3. **Fax Boxes → + Fax Box**: name, DID, destination email (inbound faxes arrive via T.38 as a PDF).

Use it - Hold music plays wherever the class is referenced; uploaded prompts play in IVRs/announcements. Generate prompts from text instead of recording them with **TTS Prompt Studio** (Part 2, §17). Invalid audio uploads are rejected.

24. Softphones

- **NeuraPBX Softphone** — native SIP softphone for **Windows** (`softphone/`, build with `build.ps1`) and **macOS** (Avalonia build). Configure: server = your PBX IP, port 5060, extension + password from the directory.
- **Web Phone** — the in-dashboard browser softphone (§14).

25. Analog Devices & Gateways (ATA / FXS / FXO)

Dashboard: Extensions (§1) and Trunks (§7). An analog telephone adapter (ATA) or analog gateway connects ordinary analog equipment — a corded phone, a fax machine, a door intercom, an overhead pager — or an existing copper phone line to NeuraPBX. Any standards-based SIP adapter works (Grandstream HT-series, Cisco/Linksys SPA, Patton SmartNode, AudioCodes MP-series, Yeastar TA-series and others). Vendors label the settings differently; the concepts below are the same on all of them, so match on meaning rather than on the exact wording.

First, identify your ports

Port	Usually coloured	What plugs in	How NeuraPBX sees it
FXS	green	an analog phone or fax machine	an **extension** — it registers like any SIP phone
FXO	red	a copper line from the phone company	a **trunk** — an outside line

A device may have one type, the other, or both. They are configured separately and neither depends on the other.

A. An FXS port as an extension

1. **Extensions → + New Extension**. Note the number and the SIP password you set.
2. In the adapter's own web interface, on its phone/FXS port page:

Setting (common labels)	Value
SIP Server / Registrar / Proxy	your PBX's IP address
SIP port	`5060`
SIP User ID / Auth ID / Username	the extension number
Password / Auth password	the extension's SIP password
Register / Registration	enabled
NAT traversal	**off** — the adapter is on your LAN
Preferred codecs	G.711 (PCMU / PCMA) first

3. Save and apply. The adapter should report **Registered**, and the extension appears online in **Extensions**.

B. An FXO port as a trunk

An FXO port needs both directions configured, and they are independent — one working does not imply the other.

Give the adapter a fixed address first. Set a static IP on the device or a DHCP reservation on your router. The trunk stores this address; if DHCP moves the adapter, the trunk stops working silently.

Outbound — PBX out to the line

1. **Trunks → + New Trunk**.
2. **Host** = the adapter's IP address. **Port** = the adapter's *FXO* SIP port (this is often `5062`, because the FXS port already uses `5060` — check the adapter).

3. **Register** = **No**. The adapter is a peer on your own network, not a provider you log in to, so leave username and password empty.
4. **Codecs** = `PCMU,PCMA`. A copper line carries nothing wideband, so G.722 and Opus have nothing to offer.
5. **Line type** = **Analog / FXO gateway** (see *Why line type matters* below).
6. **Outbound Routes** (§8) → add a route that uses this trunk. Giving it an outside-line prefix (for example, dial `0` first) keeps landline calls clearly separate from internal extension numbers.

Inbound — the line in to the PBX

7. On the adapter's FXO page, set its SIP server to your PBX's IP address on **port 5080**.
8. Turn **registration off**, and enable *outgoing/incoming calls without registration* if the adapter offers it.
9. Find the setting that decides **where an incoming line call is sent** — commonly *Unconditional Call Forward to VOIP*, with its own User ID, SIP server and destination port. It is often on a general/basic settings page rather than the FXO page (see the note below). Point it at your PBX on **port 5080** and give it a number you will route on, for example `8001`.
10. **Inbound Routes** → add a route whose **DID** is that number, pointing at whatever should answer: an extension, ring group, queue or IVR.

> **Check that your adapter can forward at all — not every one can.** Two designs are common, and they are not interchangeable:
>
> - **Unconditional forward.** The adapter answers the line and immediately calls a number you configured, so the PBX can ring reception, a group or an IVR. This is what step 9 sets up, and it is what you want for a business line.
> - **Two-stage dialling.** The adapter answers the line, plays a dial tone, and waits for the caller to key in an extension — a DISA, effectively. The caller reaches nobody unless they know a number to dial.
>
> **The setting is often not on the FXO page.** On a **Grandstream HT813** it lives under **BASIC SETTINGS**, near the bottom, as *Unconditional Call Forward to VOIP* — three boxes reading `User ID @ Sip Server : Sip Destination Port`. Nothing on the FXO page hints that it exists, and the FXO page's own help text ("answers to accept VoIP number") describes the two-stage fallback you get when it is left blank, which makes it easy to conclude the adapter cannot forward at all. It can. Verified on hardware.
>
> **Watch that third box.** The destination port defaults to **5060**, which sends the forwarded call to the extension profile, where it is challenged for a password the adapter does not have. It must be **5080**. Getting the first two boxes right and leaving the port at its default produces a gateway that looks configured and never delivers a call.
>
> If your adapter genuinely has no forward setting, its remaining option is usually *PSTN Ring Thru FXS*, which rings the analog phone on the FXS port directly and **bypasses the PBX entirely** — no routing, no IVR, no call records.
>
> **Port 5080 is the step most often got wrong.** NeuraPBX listens on two SIP ports: **5060** for extensions and **5080** for trunks. Calls from a gateway must arrive on **5080** to enter trunk routing and match your inbound routes. Sent to 5060 they are treated as an extension trying to register, and rejected.

Why "line type" matters

A SIP provider reports failure as a signalling code the PBX can read, so NeuraPBX generates a ringing tone for the caller and replaces it with a spoken message if the call fails. **A copper line has no signalling at all** — busy, reorder and "this number has been disconnected" exist only as tones you can hear on the line itself. Setting a trunk's **Line type** to **Analog / FXO gateway** tells NeuraPBX to pass the line's own audio through untouched, so callers hear the real busy tone. Left on **SIP carrier**, a caller dialling a busy landline hears a ringing tone that never stops. Leave it on **SIP carrier** for ordinary SIP trunks.

Adapter settings that are easy to miss

These are the ones that cause working-looking setups to fail in practice:

Setting (common labels)	Set to	Why
One-stage vs two-stage dialling ("stage method")	**one-stage**	Two-stage makes the adapter seize the line and hand you a dial tone instead of dialling the number the PBX sent — calls connect but never reach anyone.
Disconnect detection (current/battery reversal, busy-tone detection, polarity reversal)	enable whichever your telco uses	Without it the call stays up after the other end hangs up, and the line stays busy.
Number of rings before answering	2	Higher values just delay every inbound call.
Ring-through to the FXS port	**off**	Otherwise an incoming line call rings the analog phone directly *and* through the PBX.
Wait for dial tone	on	Prevents the adapter dialling into silence on a slow line.
Impedance / AC termination	your country's setting	A mismatch is the usual cause of echo on analog calls.

Fax on an analog port. Use G.711 (PCMU) only, disable silence suppression and comfort noise, enable T.38 if both ends support it, and limit the extension to one concurrent call. Even configured correctly, fax over IP is less tolerant of jitter than voice — for business-critical faxing, keep the machine on a direct line.

Troubleshooting

Symptom	Usual cause
Adapter never registers	Wrong SIP server or port, wrong extension password, or NAT traversal left enabled on a LAN device.
Inbound line calls never arrive	Adapter is sending to port 5060 instead of **5080**; or its forward-to-VoIP number does not match any inbound route DID; or its forward-to-VoIP destination port is still the default 5060 instead of 5080; or the forward is unset, so the adapter answers and waits for the caller to dial digits (see the note above).
Outbound call reaches the line but nothing is dialled	Two-stage dialling still set on the adapter.
Caller hears ringing forever on a busy number	Trunk **Line type** is not set to *Analog / FXO gateway*.
Call stays up after the far end hangs up	Disconnect detection not enabled, or set to the wrong method for your telco.
Echo on analog calls only	Impedance / AC termination set for the wrong country.
Adapter shows the FXO port as "not connected"	No line plugged into the FXO socket, or the line is dead — check it with an analog handset.

A peer trunk (Register = No) is expected to show as unregistered on the adapter and in trunk diagnostics; that is normal and not a fault. What matters is that the trunk shows reachable and that calls pass.

Part 2 — AI Features

NeuraPBX includes **18 toggleable AI capabilities**, managed from **AI Feature Control**. Each is enabled individually (and may be license-gated). AI requires a license with an **AI bundle and seats** (§22).

Turning AI on

1. Open the **AI Features** page (or the Dashboard's AI Feature Control widget).
2. Turn the **master switch ON** to unlock the features (ON only *unlocks* — it never auto-enables anything). Master OFF stops and locks every feature.
3. Click a feature's toggle to switch it **on**; wait for the confirmation toast.
4. If a feature is license-gated and your plan doesn't include it, the toggle is blocked — check **System → License**.
5. Enabled features react to live call events immediately — **no restart**. The **Live AI events** feed shows activity (copilot suggestions, biometric verdicts, voicemail transcripts, CRM syncs) in real time.

Choosing your AI engine (AI Features → AI Models)

Run AI in the **cloud** or **100% on-prem**, applied within ~15 s:

- **Language model (LLM)** — analysis, Copilot, QA, insights. Pick **OpenAI (cloud)** (set `OPENAI_API_KEY`) or **Ollama** locally: an **external Ollama host**, or **embedded in the AI container** (the model — Gemma, Llama, Qwen — downloads on first use).
- **Speech-to-Text (STT)** — **OpenAI Whisper (cloud)** or **local faster-whisper** (offline, no per-minute cost; larger models = more accurate but slower).

On-prem mode keeps every call, transcript, and voice print on your own hardware. Click **Save AI configuration** to apply.

The 18 AI capabilities — how to use each

Turn each on from **AI Features** → **Features** (master switch on first). Per-feature tuning — intent maps, QA rubric, spam threshold — lives on **AI Features** → **Tuning**; other one-time settings live under **System** → **Settings** unless noted. Tuning changes apply within ~15 seconds, no restart.

1. Natural-Language IVR

A conversational front door for inbound calls: instead of a rigid "press 1 for sales, 2 for support" menu, the caller just says why they're calling. NeuraPBX transcribes that opening sentence, an AI model classifies it into an intent (sales, support, billing…), and routes the call straight to the matching destination — so callers reach the right place faster, and you can add or rename intents without re-recording menu prompts. **Set up:** enable it; on **AI Features** → **Tuning** map each intent to a destination (e.g. `sales → 5001`, `support → 5002`, `billing → 5003`) and edit the greeting the caller hears (`nl_ivr_prompt`); then on **Inbound Routes** set the route's **Destination Type** to **AI IVR (natural language)**. The route's destination is the **fallback** — where the caller is sent if they say nothing, the intent has no mapping, or the AI service is unavailable — so set it to a real person or queue. **Use it:** the caller hears the greeting, says why they're calling, and is transferred. The detected intent and what the caller said appear on the dashboard. Callers whose intent can't be matched land on the fallback rather than in silence.

2. Voicemail → Email Transcription

Turns every voicemail into a readable email so no one has to dial in and listen. When a message is left it's transcribed to text, the caller's name is extracted if spoken, and the transcript plus caller ID, timestamp, and the audio file are emailed to the mailbox owner within seconds — so staff can triage and reply straight from their inbox instead of navigating a phone menu. **Set up:** enable it; confirm SMTP under **System** → **Settings** → **Email**. **Use it:** every new voicemail is transcribed, the caller's name is extracted when spoken, and the transcript + caller details are emailed to the box owner shortly after the message is left.

3. Call Transcript → Translate → Email

Produces a written, optionally translated record of a live call. The call is recorded in stereo (each party on its own channel), both speakers are transcribed separately, the text is translated into the extension owner's language, and the full transcript is emailed at hang-up. Useful for record-keeping, cross-language calls, and reviewing exactly who said what — on every call for chosen extensions, or on demand mid-call via a feature code. **Set up:** enable it and choose the extensions it applies to. **Use it:** those calls are recorded in stereo, each speaker transcribed, the text translated into the extension's language, and the transcript emailed at hang-up — automatically, or on demand with the in-call transcribe feature code.

4. Agent Copilot (Real-Time Assist)

A live assistant that listens alongside the agent and helps in real time. As the call happens it transcribes the conversation, understands the topic, and pushes suggested answers, relevant Knowledge Base snippets, and draft follow-up emails to the agent's screen — without touching the audio the caller hears. New or busy agents get expert answers on screen instead of putting callers on hold to look things up. **Set up:** enable it (and the Knowledge Base, #18, to power article suggestions). **Use it:** during a live call the AI transcribes in real time and pops suggested answers, KB snippets, and draft emails onto the agent's dashboard — without disrupting call audio.

5. Voice Biometrics Verification

Preview. Matches a caller's voice against a stored voice print, as **one signal among several** - not proof of identity. You first enrol a trusted “voice print” from a known-good call; on later calls the AI compares the live voice to that print in a few seconds and marks the caller Verified or Unverified, setting a `voice_verified` variable your IVR/dialplan can act on.

> **Limitation.** Matching compares how a voice *sounds*. It cannot tell a live speaker from a recording of that speaker, and a modern voice clone can pass. There is no replay or synthesis detection. Never let `voice_verified` alone authorise a sensitive action — pair it with an account question, an OTP, or agent judgement. **Set up:** enable it; enrol a caller's voice print from a known-good call. **Use it:** on later calls the caller is matched in ~5 seconds; the dashboard shows Verified/Unverified and a `voice_verified` variable is set on the call so your dialplan/IVR can gate sensitive actions (e.g. skip security questions only when verified). Non-enrolled or mismatched voices are flagged.

6. Speech-to-Speech Translation Bridge

Preview. An interpreter for calls where the agent and caller don't share a language. In both directions the AI transcribes what one person says, translates it, and re-speaks it in the other's language, so each hears the other in their own tongue.

> **Limitation.** Interpretation is **consecutive, not simultaneous**: each side hears the translation a few seconds after the other stops speaking, which suits structured question-and-answer rather than natural conversation. It also needs stereo call recording, and a speech voice for the target language — English and Spanish ship with the system; other languages need an OpenAI API key or an added piper voice. Because speech recognition feeds translation, a misheard word becomes a confidently spoken wrong translation. **Set up:** enable it. **Use it:** the caller and agent are interpreted live in both directions — the caller's speech is transcribed, translated, and re-spoken to the agent, and vice-versa, so each hears the other in their own language.

7. AI Summarizer + CRM Sync

Eliminates manual after-call wrap-up by writing the notes for you and filing them in your CRM. When a call ends, the recording is transcribed and an AI writes a concise summary plus action items, then pushes them — with the caller's number and call details — into the contact's record in HubSpot, Salesforce, Zoho, or any system reachable by webhook. Agents skip 5–10 minutes of typing per call and every interaction is logged consistently. **Set up:** enable it; on **System** → **Integrations** set the **CRM webhook URL** (your HubSpot / Salesforce / Zoho endpoint) and, if the endpoint needs it, a **CRM API key**. **Use it:** after each call the transcript, summary, and action items are generated and pushed to the CRM record — no manual wrap-up (saves 5–10 min per call).

8. Auto QA Scoring

Quality-assures 100% of recorded calls instead of the few a supervisor can sample by hand. Every recording is scored by AI against a rubric — greeting, compliance, empathy, and resolution by default, or your own — producing a score, a per-criterion breakdown, and a short summary. Managers get consistent, objective scoring across all agents and can spot coaching needs at a glance. **Set up:** enable it; optionally paste your own scoring rubric on **AI Features → Tuning** (leave blank for the built-in default). **Use it:** recorded calls are scored on greeting, compliance, empathy, and resolution; review scores, breakdowns, and summaries on the **QA Scores** tab.

9. Compliance & PCI Redaction

Reviews every call for compliance and protects sensitive data in the written record. After each call the transcript is scanned for script deviations and forbidden phrases, and any card number or government ID spoken aloud is masked in the stored transcript — the card is reduced to its last four digits so an agent can still reconcile a payment. Flagged calls are recorded as an exception list (clean calls are not stored), raise a dashboard alert, and are readable via **AI Tools → Compliance**. This helps meet disclosure and PCI-DSS obligations, though it complements rather than replaces a full compliance program. **Set up:** enable it. Calls must be recorded for there to be anything to scan. **Use it:** review flagged calls after the fact — each entry shows the phrases hit and whether a card number or ID was redacted. **Note:** scanning runs *after* the call, not during it, and the **audio recording is never altered** — only the transcript is masked. The recording is your record of the call and may be evidence, so it is left intact. If your obligations require the digits never to be captured in audio at all, take payments on a separate line or pause recording via the agent's phone.

10. Talk Analytics & Diarization

Breaks each recorded call down into coaching metrics. Diarization separates the audio by speaker, then the AI measures talk/listen ratio, interruptions, dead air, and speaking pace for each side. Supervisors use these to coach habits a transcript alone won't reveal — talking over customers, long silences, or rushing — surfaced on the dashboard after the call. **Set up:** enable it. **Use it:** after each recorded call you get a speaker-separated transcript plus talk/listen ratio, interruptions, dead air, and speaking pace, surfaced on the dashboard.

11. Conversation Insights & Predictive CSAT

Mines the content of every call for trends you'd otherwise miss. Each conversation is auto-tagged with its topics and disposition, and the AI predicts a satisfaction (CSAT) score from the language used — no survey needed, so you get a read on every call, not just the few that respond. The Insights tab aggregates top topics, root causes, average predicted CSAT, and volumes over time. **Set up:** enable it. **Use it:** each call is auto-tagged with topics and dispositions and given a predicted CSAT (no survey needed). Aggregate topics, average predicted CSAT, and volumes appear on the **Insights** tab.

12. Spam Screening

Preview. Scores each inbound call for spam likelihood before an agent is disturbed, using only what this PBX already knows — no subscription and no internet needed. **STIR/SHAKEN** is used where your carrier signs calls and passes the result — a *failed* validation means the caller ID was forged (a strong signal), attestation **C** means the carrier cannot vouch for the number, and attestation **A** means it verified the caller owns it, which also switches off the spoofing heuristic below. Most trunks send nothing, and that absence is treated as neutral. Alongside it, five local signals are combined: your **blacklist**, a **withheld/anonymous** caller ID, one number reaching **several different extensions** in a short window (a dialler walking your range), a history of **repeated**

very short calls, and **neighbour spoofing** (a caller ID faked to share your own number's prefix). If the combined score meets your threshold the call is flagged (`spam_suspected` , plus `spam_score`) so you can send it to voicemail or a challenge, while legitimate callers pass through untouched. Every score is logged with the reasons that produced it, so you can always answer why a call was diverted. **Set up:** enable it; set the spam **threshold** slider (0-1, default 0.9) on **AI Features -> Tuning**. **Use it:** inbound calls are scored on pickup and flagged calls can be diverted.

> **Limitation.** There is no carrier reputation lookup (that needs a paid feed) and **no synthetic-voice/deepfake detection**. STIR/SHAKEN is the only signal that can judge a **first-time** caller, and it only works if your carrier signs calls and forwards the result; without it, an unknown number from a clean-looking caller ID scores zero until it establishes a pattern here. Do not rely on this as your only spam defence. **Set up:** enable it; set the spam **threshold** slider (0–1, default 0.9) on **AI Features → Tuning**. **Use it:** inbound calls are scored on pickup; a score at/above the threshold flags the call (`spam_suspected`) so you can divert it to voicemail or a challenge before an agent is disturbed. Normal calls pass through untouched.

13. Language Auto-Detection Routing

Detects the language a caller is speaking within the first few seconds and routes them accordingly. Instead of "for English press 1, para español oprima 2", the AI listens to the opening words, identifies the language, and transfers the call to the queue, agent, or bot you've mapped to it — so callers who don't share your default language reach a capable handler without menu friction. **Set up:** enable it; on **AI Features → Tuning** map each detected language to a destination (e.g. `es → 5001` , `fr → 5002`) and make sure those destinations exist. **Use it:** the caller's language is detected in the first seconds and the call is transferred to the mapped queue or bot.

14. Recording Noise Reduction

AI Features page. Cleans background noise out of call **recordings** before they are transcribed, so transcription, Auto QA, Conversation Insights and Voicemail-to-Email all read clearer audio. Useful where accuracy is suffering: noisy rooms, shop floors, hands-free callers. **Set up:** enable **Recording Noise Reduction** on the AI Features page. **Use it:** nothing else to do — every transcription from then on runs on a cleaned copy.

> **Your recordings are never modified.** A temporary cleaned copy is transcribed and then deleted; the stored recording is untouched, because it is your record of the call and may be evidence. If cleaning fails for any reason the transcription simply runs on the original.
>
> **This is not real-time.** It does not change what anyone hears during a call. Live suppression would need a media filter that the bundled FreeSWITCH build has no engine for and cannot compile without licence-gated development headers, so it is not offered rather than offered as a switch that does nothing.
>
> Leave it off unless you need it: it costs CPU on every transcription and gains nothing on already-clean audio. Stereo is preserved, so speaker separation and Talk Analytics are unaffected.

15. AI Outbound Campaigns

Preview. An auto-dialer for proactive outreach - appointment reminders, satisfaction surveys, payment reminders. You define a campaign (numbers, caller ID, and a script) and NeuraPBX places the calls, tracking progress per campaign. Note: outbound dialling is heavily regulated (TCPA/prior consent, DNC scrubbing, calling hours) - confirm compliance before running one. **Set up:** enable it, then on the **Campaigns** tab create a campaign - name, type, comma-separated numbers, caller ID. **Use it:** press **Start**; calls are placed and progress shows per campaign.

> **Limitation.** Dialling, the calling-hours window and DNC/opt-out scrubbing all work. **Answering-machine detection, voicemail drop and the AI agent leg do not** - a call that is answered currently hears silence. Treat this as a dialler, not an AI outbound agent. **Set up:** enable it, then on the **Campaigns** tab create a campaign — name, type, comma-separated numbers, caller ID, and the AI agent script. **Use it:** press **Start**; calls are placed to the contacts with answering-machine detection and voicemail drop; the AI script plays and responds. Progress shows per campaign.

16. Omnichannel AI Bot

Preview. One AI assistant answering customers consistently across text channels, not just voice. Inbound messages POSTed to NeuraPBX's webhook are answered by the same LLM that powers the voice bot, grounded in your Knowledge Base, with per-correspondent conversation memory so a thread keeps its context. **Set up:** enable it; POST inbound messages to `/api/ai-tools/omnichannel/webhook` with `channel`, `from` and `message`. **Use it:** the endpoint returns the reply for you to deliver.

> **Limitation.** NeuraPBX generates the replies but **does not send or receive anything**. No SMS, chat or email transport ships - you must connect your own provider to post messages in and deliver replies back out. **Set up:** enable it; point your SMS/chat/email channels at `/api/ai-tools/omnichannel/webhook`. **Use it:** inbound messages are answered by the same LLM that powers the voice bot, using your Knowledge Base, with shared context across channels.

17. TTS Prompt Studio

Creates professional IVR greetings and announcements from typed text — no voice-over artist or recording booth needed. Type the wording, pick a voice, and Generate; the AI synthesizes clean speech and saves it straight into System Recordings, ready to attach to any IVR, queue announcement, or hold message. Updating a prompt is as fast as editing text, and every prompt keeps a consistent brand voice. **Set up:** enable it, then open the **TTS Studio** tab. **Use it:** type your greeting/announcement, choose a voice, and **Generate** — the audio is saved straight into **System Recordings**, ready to attach to IVRs and Announcements.

> **Offline by default.** A local speech engine (piper) ships inside the container with **English and Spanish** voices, so this works with no internet and no API key. If an OpenAI key is configured it is used instead, giving a wider choice of voices.
>
> **The limit is language coverage, not capability.** A language other than English or Spanish needs either an OpenAI key or a piper voice you supply yourself — mount the `.onnx` file and point `PIPER_MODEL` at it.

18. Knowledge Base (RAG)

The library of company knowledge that grounds the AI's answers so they're accurate rather than guessed. You upload your FAQs, policies, and product docs; when Agent Copilot or the Omnichannel Bot answers a question it retrieves the most relevant passages and answers from them — a technique called RAG (retrieval-augmented generation). In-scope questions get correct, sourced answers and out-of-scope ones are handled gracefully instead of fabricated; use Test Retrieval to preview what the AI would pull for a sample question. **Set up:** enable it, then on the **Knowledge Base** tab add your FAQs, policies, and product docs (paste or name them). **Use it:** use **Test Retrieval** to preview answers to a sample question. The KB powers **Agent Copilot** (#4) and the

Omnichannel Bot (#16); in-scope questions get accurate, sourced answers, and out-of-scope ones are handled gracefully rather than fabricated.

The AI Features page & on-demand analysis

The **AI Features** page groups the tooling above into tabs — **Features** (toggles), **AI Models** (engine choice), **Knowledge Base**, **QA Scores**, **Insights**, **Campaigns**, and **TTS Studio**. Separately, the **AI Assistant** page runs on-demand analysis of any recorded call: transcript, sentiment, summary, action items, and keywords.

Appendix — Star-code quick reference

All codes are editable on the **Feature Codes** page (§9); changes regenerate the dialplan instantly.

Code	Feature
`*97`	Voicemail check (own box, PIN)
`*98`	Voicemail direct access
`*72` + number	Enable call forwarding
`*73`	Disable call forwarding
`*78` / `*79`	DND on / off
`*8`	Call pickup (your group)
`*43`	Echo test
`411`	Dial-by-name directory
`*0` + code	Speed dial
`*54` + ext	Intercom (auto-answer)
`*33` + ext	Supervisor monitor (1=spy · 2=whisper · 3=barge)
`*1`	In-call: blind transfer — press `*` `1`, dial the destination + `#`
`*2`	In-call: attended transfer — press `*` `2`, dial destination + `#`, announce, hang up
`*280`	Day/Night override toggle
`*281` … `*289`	Named call-flow toggles
`*68`	Schedule a wake-up call

For employees: `*97` voicemail · `*78` / `*79` DND · `*8` pickup · `*0` +code speed dial · **My Portal** for visual voicemail and Follow Me. **For receptionists:** park a call then dial its slot to retrieve · page a group · transfer from the Operator Panel. **For supervisors:** Monitoring page or `*33` +ext → DTMF 1 spy / 2 whisper / 3 barge · Reports wallboard. **For admins:** AI toggles on the AI Features page · routing under Time Conditions + Outbound Routes · `*280` / `*281…` flow switches · backups, tokens, and the audit log on System.

Appendix — Legal & Compliance

NeuraPBX is self-hosted software that **you** configure and operate, so using it lawfully is your responsibility. Telecom, call-recording, marketing, biometric, and privacy laws vary by country, state, and industry and change over time. This is an operator checklist, **not legal advice** — consult your own counsel. NeuraPBX ships safe defaults, but you must still configure each feature for your situation.

Per-feature responsibilities

- **Call recording / transcript / Auto QA / Talk Analytics / Voicemail→Email** — obtain any required consent before recording, transcribing, or monitoring (many jurisdictions require **all-party** consent). The "this call may be recorded" announcement is **on by default** (`recording_consent_announcement`, System → Settings). It notifies the recorded extension; for **external inbound** calls, also add a recorded "may be recorded" notice to your inbound IVR/greeting so the outside caller hears it.
- **AI Outbound Campaigns** — **TCPA**: prior express consent, Do-Not-Call / opt-out scrubbing, permitted calling hours, and honest caller ID. Creating a campaign now **requires a consent attestation**; set the **DNC list** and **calling-hours window** (default 8:00–21:00, local server time) on **AI Features → Tuning**. Campaigns never dial numbers on the DNC list and won't run outside the window.
- **Voice Biometrics** — provide notice and obtain **written consent** before enrolling or matching voiceprints, and keep a retention/destruction policy where required (e.g. Illinois **BIPA**). Do not enable silent enrolment.
- **Emergency calling (E911)** — **Kari's Law** (911 dialable with no prefix) and **RAY BAUM'S Act** (dispatchable location). Configure an emergency Outbound Route + E911 location; verify 911 cannot be blocked by a PIN set or allow-list; **test it with your carrier**.
- **Supervisor Monitoring (spy / whisper / barge)** — wiretap and employee-monitoring laws. Disclose monitoring to agents and follow your jurisdiction's notice/consent rules.
- **Spam Screening** — don't block legitimate or emergency calls, and don't defame a caller by labeling. Keep the threshold conservative, whitelist known-good numbers, and never divert 911.
- **Compliance & PCI Redaction** — reduces exposure but is **not** a substitute for a full PCI-DSS / compliance program, and does not make your use "PCI compliant." Treat it as an aid.
- **Omnichannel Bot / Agent Copilot / Knowledge Base** — you are **bound by your bot's statements**, and AI output can be wrong. Add disclaimers, guardrails, and human handoff; don't use it to give regulated (legal / medical / financial) advice.
- **CRM Sync & stored call data** — **GDPR / CCPA**: provide notice, secure and encrypt data, set retention limits, and honor deletion / data-subject requests. Put a data-processing agreement in place with any CRM you push to.
- **Translation Bridge** — mistranslation risk on medical/legal/emergency calls; use a human interpreter for critical conversations.
- **DISA / SIP trunks** — toll-fraud risk; use PINs, destination allow-lists, and the built-in toll-fraud + fail2ban limits.

Safe defaults NeuraPBX ships

- Recording-consent announcement: **on**.
- Outbound campaigns: consent attestation **required**; DNC scrub and calling-hours window (default **8:00–21:00**) **enforced**.
- Toll-fraud + fail2ban protection: **on**.