NeuraPBX — User Manual

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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. `https://<server-ip>/`) 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 (with optional per-seat AI bundles) 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).

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 (Yealink, Grandstream, Polycom, Snom, Cisco, Fanvil) with their config and BLF keys.

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** for the provisioning URL.
5. Point the phone's auto-provision server at `http://<your-pbx>/provision/<MAC>.cfg` (or 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. `*32` blacklists the last caller (blocked callers get 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, AI minutes), 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, with optional per-seat **AI bundles** (Essentials / Agent Pro / Ultimate). **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 additionally needs an AI bundle with seats. 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).

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`); point an inbound route at the NL-IVR.

Use it: the caller says why they're calling; NeuraPBX classifies the intent and transfers the call. The detected intent appears on the dashboard.

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

Confirms a caller's identity from the sound of their voice, as an added security layer. 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 about five seconds and marks the caller Verified or Unverified, setting a `voice_verified` variable your IVR/dialplan can act on — for example skipping security questions only when the voice matches, and flagging mismatched or unenrolled voices for review.

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

A live 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 in near real time — so each hears the other in their own tongue. It removes the need for a human interpreter or a separate language queue for many routine conversations.

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

Monitors calls live for compliance and automatically protects sensitive data. As the call is transcribed, the AI watches for script deviations and forbidden phrases and raises a dashboard alert when they occur; when a caller reads out a card number or SSN, it masks those seconds of the recording and redacts the digits from the transcript. This helps meet disclosure and PCI-DSS obligations, though it complements rather than replaces a full compliance program.

Set up: enable it.

Use it: calls are monitored live — script deviations and forbidden phrases raise a dashboard alert, and when a card number or SSN is spoken the recording auto-pauses (masks) and the transcript is redacted.

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 & Deepfake Screening

Screens inbound calls for robocalls, spam, and AI-cloned (deepfake) voices before they reach an agent. On pickup the AI scores how likely the call is to be unwanted; if it meets your threshold the call is flagged (`spam_suspected`) so you can send it to voicemail or a challenge instead of ringing a person, while legitimate callers pass through untouched — cutting the time agents lose to junk calls.

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. Real-Time Noise Suppression

Cleans up call audio in real time so background noise doesn't derail conversations. Using RNNoise on bridged calls, it strips fans, traffic, keyboard, and café chatter from both sides while leaving speech clear. The result is more intelligible calls and better transcription/AI accuracy, with nothing to configure.

Set up: just enable it — no configuration.

Use it: background noise (fan, traffic, music) is filtered (RNNoise) from both agent and caller audio on bridged calls, while speech stays clear.

15. AI Outbound Campaigns

An AI-driven auto-dialer for proactive outreach — appointment reminders, satisfaction surveys, and payment reminders. You define a campaign (numbers, caller ID, and an AI script) and NeuraPBX places the calls, uses answering-machine detection to leave a voicemail drop when no one answers, and lets the AI voice play the script and handle simple replies, tracking progress per campaign. Note: outbound dialing 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, 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

Gives you one AI assistant that answers customers consistently across text channels, not just voice. Point your SMS, web-chat, and email channels at NeuraPBX's webhook and inbound messages are answered by the same LLM that powers the voice bot, grounded in your Knowledge Base, with context shared across channels so a conversation can move from chat to voice without starting over. Routine questions get instant, on-brand answers around the clock.

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.

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

Code	Feature
`*78` / `*79`	DND on / off
`*70`	Toggle call waiting
`*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: toggle recording
`*11` / `*12`	Hot-desk login / logout
`*66`	Camp-on (call me when a busy ext frees)
`*280`	Day/Night override toggle
`*281…*289`	Named call-flow toggles
`*32`	Blacklist the last caller
`*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 & Deepfake 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**.

Before you go live

- Confirm the call-recording consent rules for every jurisdiction you operate in, and verify the announcement reaches outside callers (via an inbound IVR notice).
- Populate the campaign DNC list and confirm consent before running any outbound campaign.
- Configure and **test** emergency (E911) dialing with your carrier.
- Publish a privacy notice and set retention/deletion for recordings, transcripts, and contact data.
- Present your Terms & Conditions / Acceptable-Use policy and require acceptance — and have counsel review it. (The download page's "I agree to the Terms" gate and the online Terms cover operator responsibilities, warranty disclaimer, limitation of liability, and indemnification.)