Navijin

ADMINISTRATOR MANUAL

NAVIJIN VBX VOICE SYSTEMS

VBX3000 IP PBX Administrator Manual

Install and configure the Navijin VBX3000 IP PBX: hardware and cabling, network, extensions and departments, analog, SIP and E1/T1 trunks, call routing, IVR, queues, voicemail, recording, conferences, feature codes, security and maintenance.

Hardware · Installation · Extensions · SIP trunks · Analog & E1 · Routing · IVR & queues · Feature codes · Security

Applies to
Navijin VBX3000 IP PBX (VBX3000-128S chassis with FXS, FXO and E1/T1 boards)

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ABOUT THIS GUIDE

About the VBX3000

The Navijin VBX3000 is a modular IP PBX and voice gateway for small and medium enterprises, hotels and campuses. It connects **IP phones and analog phones** on one system and links them to the outside world through **analog lines (FXO), SIP trunks and E1/T1 (PRI, R2, SS7)**. It includes an IVR auto attendant, call queues, voicemail to email, conference bridges, call recording, billing and a full set of feature codes, all configured from the web interface. This manual follows the order of a real project: hardware, installation, network, extensions, trunks and routing, then features, security and maintenance.

Conventions: menu paths are shown in **bold** (for example **Network** → **VLAN**); commands and values in `monospace`. Screenshots may differ slightly by firmware version.

Who this guide is for

- Telecom installers and integrators deploying the VBX3000
- IT administrators running the company phone system
- Hotel and campus operators managing extensions and lines

What you need

- A VBX3000 with the FXS, FXO and E1/T1 boards your project needs
- A 19-inch cabinet or shelf, earthing and a stable power supply (UPS recommended)
- A computer with a modern browser
- Your numbering plan, SIP trunk details from the provider, and line assignments

Factory defaults

Setting	Default
LAN address	`192.168.100.1` / 255.255.255.0 (DHCP server on)
Web interface	`http://192.168.100.1`
Administrator	`admin` / `admin` plus verification code (change it immediately)
User self-service portal	`http://<PBX-address>/users.php`, initial user password `111111`
SIP registration password	`Aa111111` (change per extension)
Voicemail PIN	`0000`
Console	RJ45, 115200 bit/s
Web access from WAN	Off

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SECTION 01

Hardware overview

The VBX3000 is a 2U modular chassis. Boards are combined to fit the number of extensions and lines.

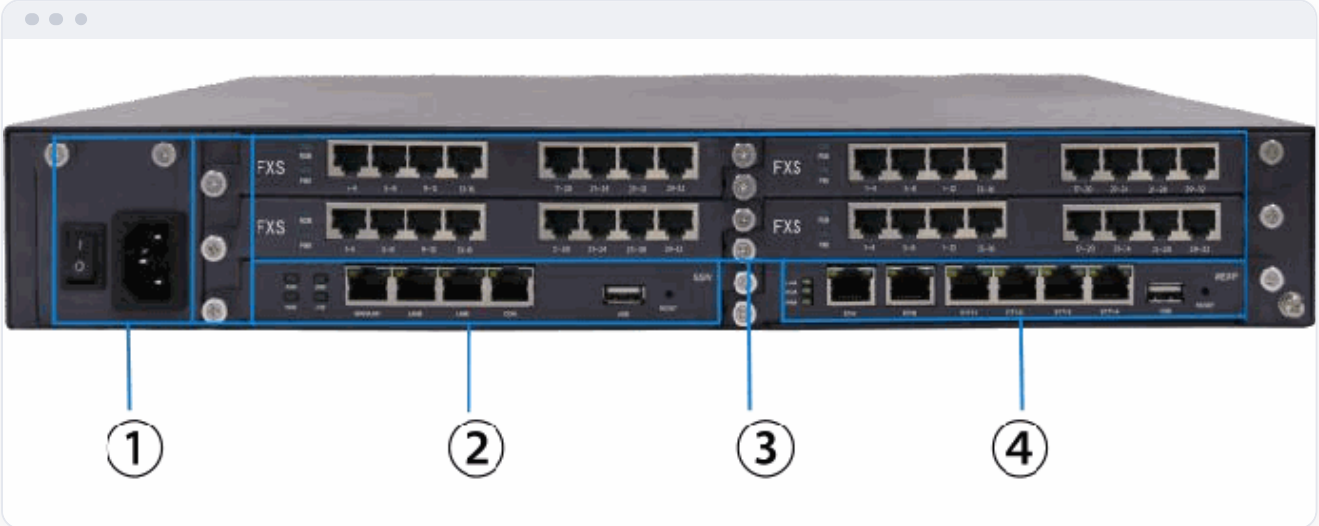


Figure 1. VBX3000 rear view with power, system and line boards.

Item	Specification
Chassis	2U metal, 6 slots, 440 × 348 × 75 mm, ≤ 9 kg
Power	210 W power board
Capacity	Up to 128 analog ports (FXS + FXO) and up to 4 E1/T1
SSW system switching board	1 × WAN + 2 × LAN (10/100/1000 Mbit/s RJ45), 1 × console RJ45 (115200 bit/s), 1 × USB 2.0
MEXP extension management board	2 × Ethernet for upgrade and maintenance, 1 × console, 1 × USB 2.0
E1/T1 board (option)	4 × E1/T1, RJ45, 120 Ω
FXS board	32 analog extensions on 8 × RJ45
FXO board	32 analog trunk lines on 8 × RJ45

RJ45 pins (FXS/FXO boards)	Line
1–2	Line 1
3–4	Line 2
5–6	Line 3

7–8	Line 4

> **NOTE**
> Each RJ45 on an FXS or FXO board carries four analog lines. Use a breakout panel or patch panel to split them into RJ11 ports.

SECTION 02

Site preparation and safety

A voice system runs 24/7. Install it in a clean, cool, earthed and protected location.

Requirement	Recommendation
Temperature and humidity	Equipment-room conditions, non-condensing; avoid direct sun and heat sources
Cleanliness	Dust-free room; dust causes overheating and contact faults
Static	Wear an earthed wrist strap when handling boards
Magnetic interference	Keep away from large transformers, motors and radio transmitters
Lightning	Surge protection on analog lines and E1 lines entering the building
Mounting	19-inch cabinet or a stable shelf that carries the weight, with space for airflow

> **IMPORTANT**
> Connect the protective earth before any other cable, and disconnect it last. Analog lines from the street can carry dangerous voltages during storms.

> **IMPORTANT**
> Switch off and unplug the PBX before you insert or remove boards.

SECTION 03

Installation and cabling

Mount, earth, power and connect in this order.

1. **Mount the chassis**

   Fix the PBX in the cabinet with rack ears and screws, or place it on a stable work surface.

2. **Connect the earth cable**

   Connect the earth terminal to the cabinet or building earth bar with the shortest possible cable.

3 Connect power

Check that the supply voltage matches the label, then connect the power cable. Use a UPS for the PBX, the PoE switch for the phones and the router.

4 Connect Ethernet

WAN to the internet router or provider; LAN to the phone network switch. Use Cat5e or better.

5 Connect analog lines

FXO ports to the operator's analog lines; FXS ports to analog phones, fax machines or door phones, through a patch panel.

6 Connect E1/T1

Connect the E1/T1 port to the provider's NTU or another PBX. Use a crossover cable (pins 1-2 ↔ 4-5) between two terminal devices, and a straight cable to a device that is already crossed. Agree the cable type with the provider.

7 Power on and check

Switch on. The self-test takes 1–2 minutes. Check that the front indicators are normal.

> **TIP**
> Label every patch cable with its extension or line number. Most later faults are found faster with good labels than with any tool.

SECTION 04

First sign-in

Configure the PBX from its web interface.

1 Connect your computer

Connect your computer to a LAN port. Leave it on automatic IP (DHCP); the PBX gives it an address. Or set a static address such as `192.168.100.10/24`.

2 Open the web interface

Open `http://192.168.100.1` in your browser and choose English.

3 Sign in

Enter `admin` / `admin` and the verification code, then click **Login**.

4 Change the password

Go to **System Management › Basic Settings** and set a strong administrator password straight away.

The menu bar at the top selects the module (Status, Network, Voice Config, QoS, Objects, Security, System). The navigation pane on the left shows the pages of the module; the settings area is on the right.

IMPORTANT

A phone system with default passwords is a prime target for toll fraud: attackers place expensive international calls through it. Change all default passwords before connecting trunks.

SECTION 05

Network

Give the PBX its LAN and WAN addresses and decide how it connects to the internet.

Page	Settings
Basic Setup › LAN Setup	LAN address and mask (default 192.168.100.1/24). Sign in again at the new address afterwards
Basic Setup › WAN Setup	Static IP, DHCP or PPPoE on the WAN port; operating mode Gateway or Bridge
Basic Setup › DHCP	DHCP server for phones on the LAN
Advanced › Static / dynamic route	Routes to other sites and networks
Advanced › NAT, Port Mapping, UPnP, ALG	Address translation and SIP ALG
Advanced › DDNS	Dynamic DNS name for a changing WAN address
Advanced › IGMP proxy / VLAN	Multicast (IPTV)
VPN	IPsec, L2TP and PPTP tunnels to branches and remote users

Operating mode	Use when
Gateway	The PBX is the router of the voice network and connects to the provider
Bridge	A router already exists; the PBX sits behind it with a management address in the LAN

TIP

In most offices the PBX sits behind the company router with a static LAN address. Put the phones in a voice VLAN with QoS so calls stay clear when the network is busy.

NOTE

Turn off SIP ALG on the company router when the PBX and SIP trunks work through NAT. A SIP ALG often rewrites packets wrongly and causes one-way audio.

SECTION 06

Quick setup wizard

The wizard sets up extensions, a trunk and an outbound route in three steps, enough for internal and external calls.

1 **Extensions**

Open **Voice Config › Quick Guide** and click **Step 1**. Create the first extensions.

2 **Trunks**

Click **Next Step** and configure an analog trunk (FXO channels) or a SIP trunk with the provider's address, port, account and password.

3 **Routes**

Click **Next Step** and create the outbound route, for example prefix `0` for external calls. Click **Finish**.

> **TIP**
> Use the wizard for a first test call, then refine users, permissions and routing on the detailed pages below.

SECTION 07

Users and extensions

Each person is a user with an account, an extension and permissions.

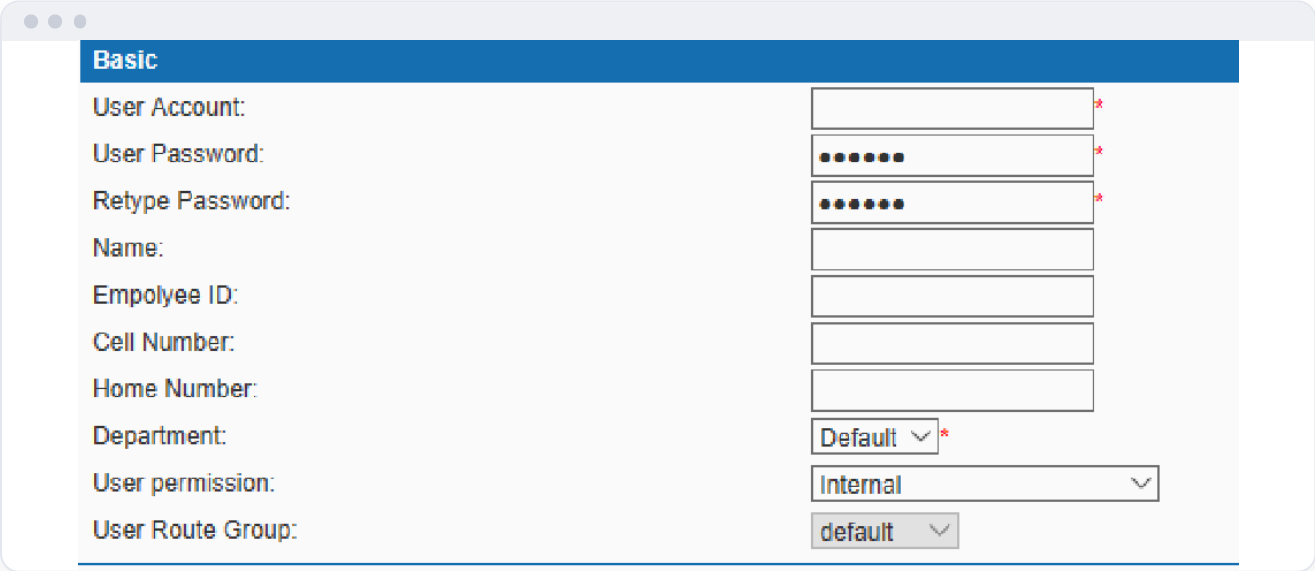


Figure 2. Add user: basic information.

Basic setting	Description
User account	Unique ID (1–20 characters); cannot be changed later; used in reports

Password	Self-service portal password, 6–20 characters (default 111111)
Name, employee ID, cell and home number	Directory information
Department	Group for permissions and reports
Route group	Which outbound routes the user can use (default: default)
VIP class	Ordinary or VIP1–VIP5 (enable VIP in Global Setting)

Extension setting	Description
Extension number	Internal number of the analog or SIP phone
DID number 1/2	External numbers that reach this extension directly
Permission	Internal, Enterprise, Local, National or International. Higher includes lower; default National
Bill type	Post-payment (default), prepayment or card user
Bind FXO channel	Use a specific outside line for this extension
Device type	Analog (choose the FXS channel) or SIP

SIP option	Default and advice
Register account / password	Default password Aa111111. Use 8–20 characters with upper case, lower case and digits
Permitted registry addresses	Up to 5 networks the phone may register from. Set them to your LAN
Max calls	2 simultaneous calls per phone
Fax mode	None, T.30 pass-through or T.38
DTMF	RFC 2833 (default), INFO or in-band
P2P media	Audio goes directly between phones. Calls with P2P cannot be recorded
Authentication	Keep it on

- **Call time limits** (minutes per call): internal 0 = unlimited, local 1440, national 480, international 120.
- **Voicemail**: enable, set the email address, PIN (default 0000) and size. SMTP must be set first.
- **Unanswered call**: hang up (default), voicemail, IVR or another extension.
- **Import/Export**: add many users at once from an Excel file, then click **Reload**.

Departments (User Config › Department) group users. For each department you can allow external calls, calls inside the group and calls to other departments.

IMPORTANT

Give international permission only to the extensions that need it. Most toll fraud uses extensions that never needed international calls.

SECTION 08

User self-service portal

Users manage their own forwarding, voicemail and status from a browser.

Users open `http://<PBX-address>/users.php` and sign in with their user account and password (initial `111111`; they must change it).

Function	What users can do
Basic info	View their details; change account and SIP passwords
Status	Absent, Do Not Disturb, phone login/logout, call waiting
Call failure	Hang up, voicemail, forward to an extension or IVR
Follow Me	Ring the extension and mobile or other numbers together or in sequence
Call forwarding	Unconditional, on busy, on no answer
Voicemail	Settings, and listen with *97 (local) or *98 (remote)
Blacklist, alarm clock, music ring	Personal call blocking, wake-up calls and ringback music
Call log	Their own calls

NOTE

When Follow Me is active, unconditional and no-answer forwarding are disabled.

SECTION 09

Trunks

Trunks connect the PBX to the provider and to other PBXs.

Trunk type	Connects to	Notes
Analog (FXO)	Analog lines from the telephone company	Select the FXO channels, permission, route group and caller ID
SIP	SIP provider or another IP PBX	Address, port, transport, account; up to 32768 calls (default 10)

PRI (E1/T1)	ISDN primary rate from the provider or another PBX	Enable PCM first; agree switch type and side
R2 (E1)	Channel-associated signalling networks	Country variant, number lengths, caller category
SS7 (E1)	Carrier networks	ITU, ANSI or China variants, ISUP, point codes from the operator

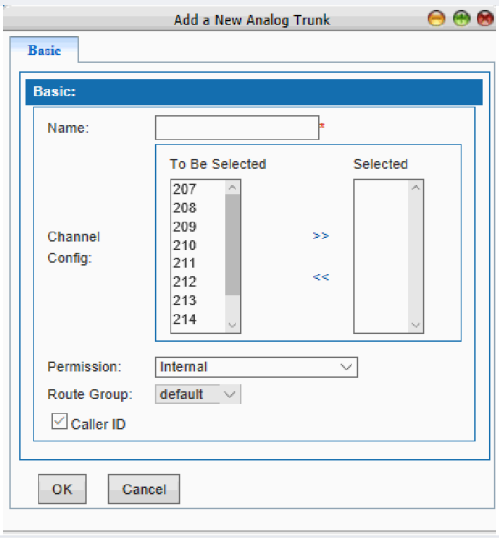


Figure 3. Add an analog trunk: name, FXO channels, permission and route group.

1 Add a SIP trunk

Voice Config › Trunks Config › Trunk Config › SIP Trunk › Add. Enter name, provider address or domain, port and transport (UDP or TCP, as the provider), permission, DID number, route group, interface type and max calls.

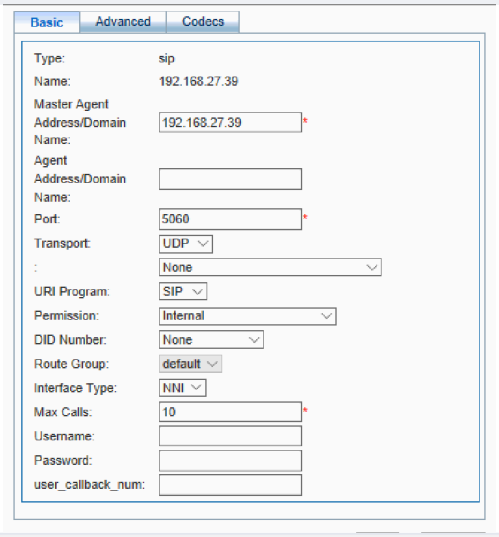


Figure 4. SIP trunk basic settings.

Interface type	Meaning
UNI	The PBX registers to the provider like a user; the account is set in **SIP Registry**

NNI	Network-to-network peering, for example PBX to PBX; IP authentication or user/password

- **Advanced**: proxy server, external IP (for NAT), OPTIONS keep-alive (60 s), encryption, DTMF (RFC 2833 default), user=phone, fax T.30/T.38.
- **Codec**: choose and order the codecs: G.711A/U, G.729, G.723, G.726, GSM, Speex, iLBC and AMR.

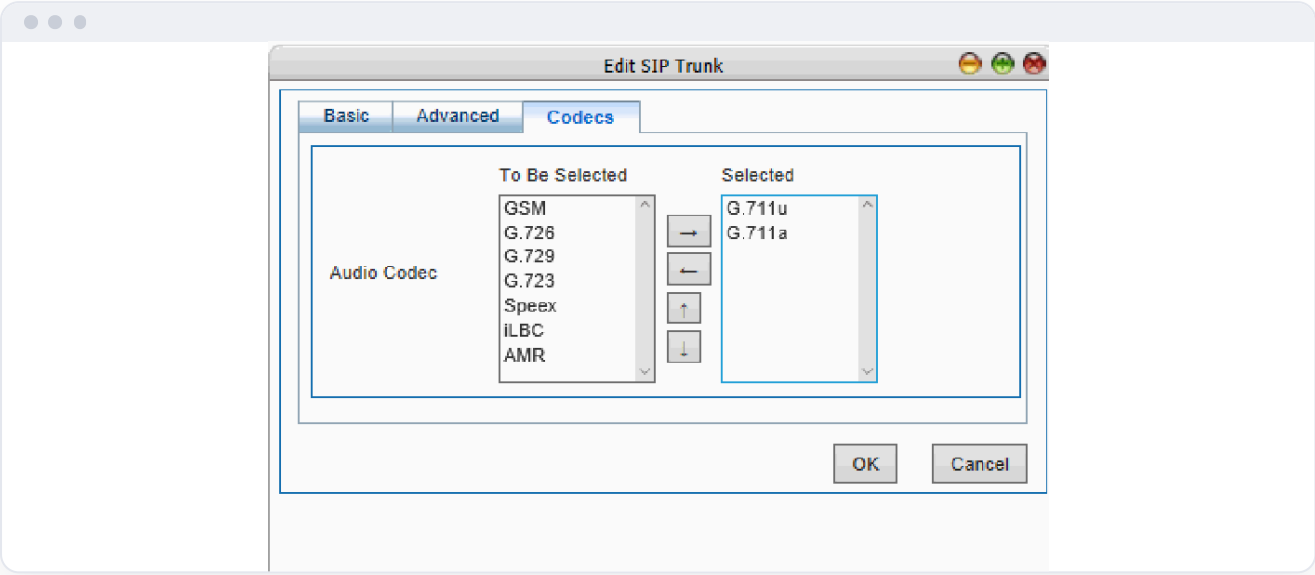


Figure 5. Codec selection and priority.

2 Register a UNI account

Trunks Config › SIP Registry › Add. Choose the provider, enter SIP user name, registry name and password, set **Registry = Yes** (default interval 1800 s) and click **Register**.

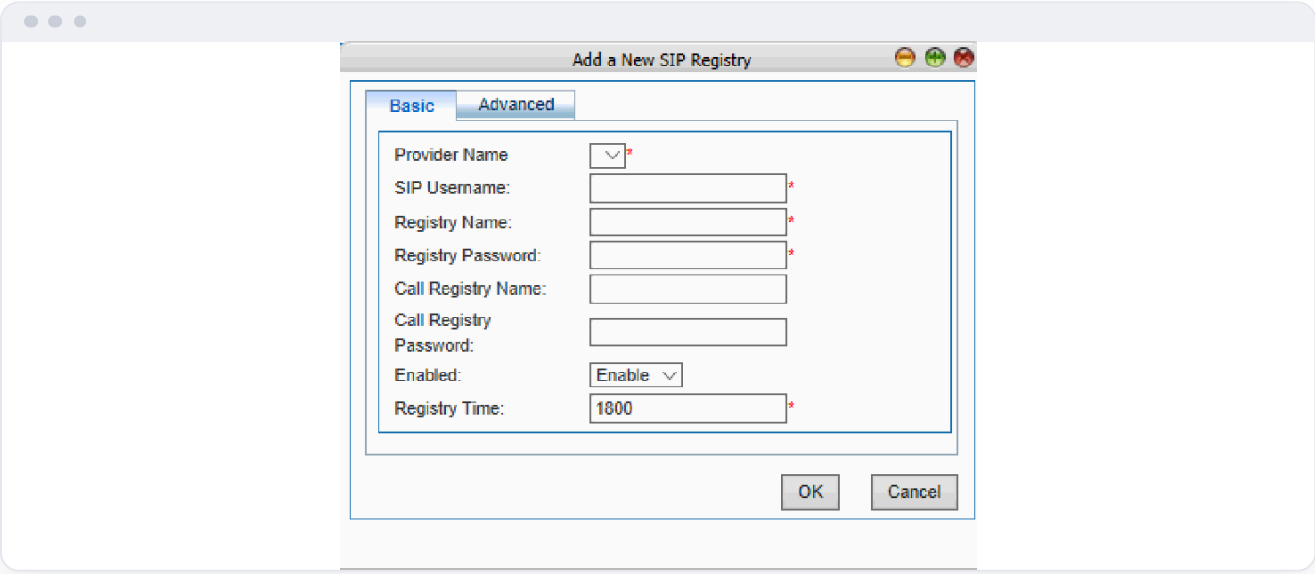


Figure 6. SIP registry account.

> **TIP**
>
> Put G.711A (Europe, most of the world) or G.711U (North America) first for best quality; add G.729 only for low-bandwidth links. Both ends must share at least one codec.

> **NOTE**
>
> For PRI, one side is network and the other user. Towards the public network the VBX3000 is normally **user** side; towards a smaller PBX it is **network** side.

SECTION 10

Inbound call routing

Decide where external calls go, by trunk and by time of day.

1 Create the route

Voice Config › Trunks Config › Inbound Call › Add. Enter the name and the days and hours when the route applies.

2 Choose the trunks

On **Trunk Group**, add the trunks this route handles.

3 Choose the action

On **Action**, send calls to an extension, the IVR or an operator.

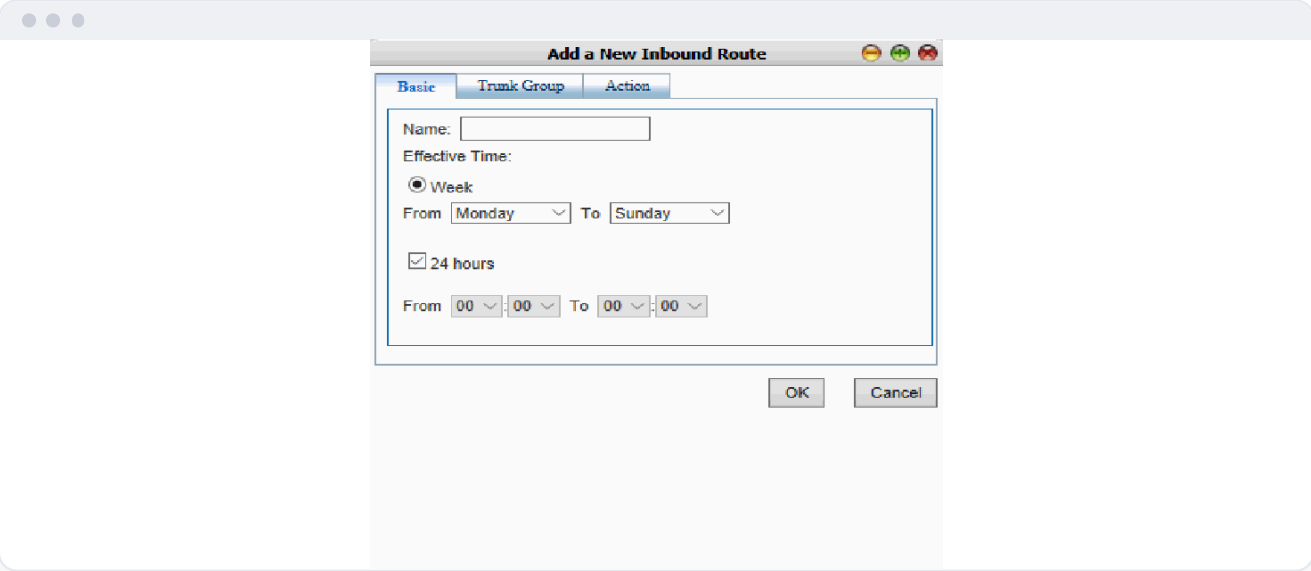


Figure 7. Inbound route with effective days and hours.

> **TIP**
>
> Create two routes on the same trunks: business hours to the IVR or reception, and out of hours to a closed message or voicemail.

DNIS routes calls by the dialled number (DID) and **CNIS** by the caller's number, for example VIP customers to a priority queue. **Number Transfer** rewrites numbers.

SECTION 11

Outbound call routing

Outbound routes choose the trunk for each type of call, with automatic fallback.

1 Basic

Trunks Config › Outbound Call › Add. Enter the name, the prefix users dial (for example `0`), the call type (enterprise, local, national, international) and the effective days.

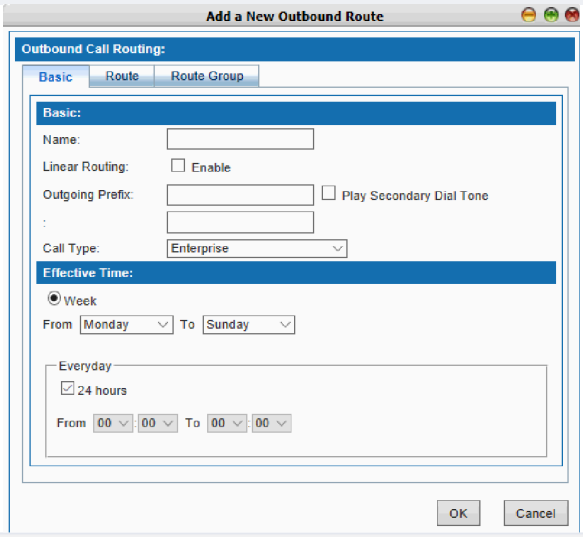


Figure 8. Outbound route: name, prefix, call type and time.

2 Route

On **Route**, add one or more trunks in order of priority. If the first is busy or down, the next is used. Set how the prefix is changed when dialling out.

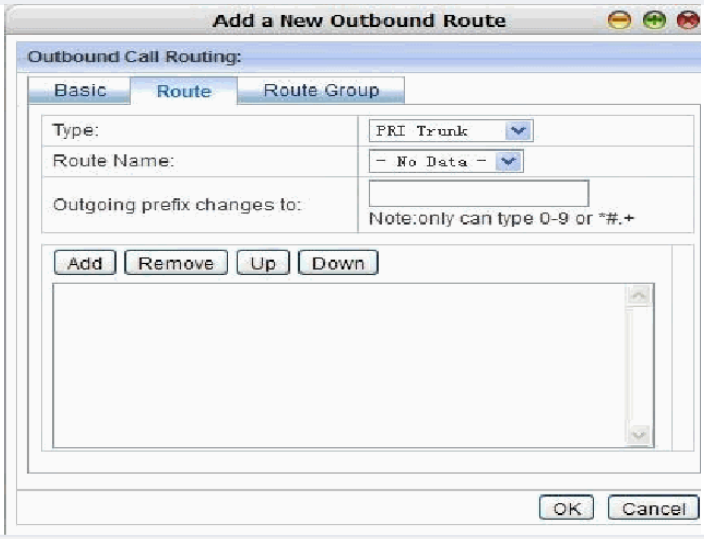


Figure 9. Trunks and their order in the route.

3 Route group

Add the route to one or more route groups. Users can only use routes in their own route group.

Example	Dial	Trunks in order
Local and national	0 + number	Analog lines, then SIP
International	00 + number	SIP trunk only, international permission required
Branch office	8 + extension	SIP NNI trunk to the branch PBX

> **NOTE**
> A call goes out only if both the user's permission and an outbound route allow it. **Linear routing** dials out without a prefix, with a second dial tone if enabled.

SECTION 12

IVR auto attendant

Greet callers and let them choose a department, with nested menus and time-based behaviour.

1 Create the IVR

Voice Config › PBX Features › IVR › Add. Enter a name and click **OK**.

2 Add the events

For each key or event, choose the action and its content, click **Add** and order the actions with **Up** and **Down**.

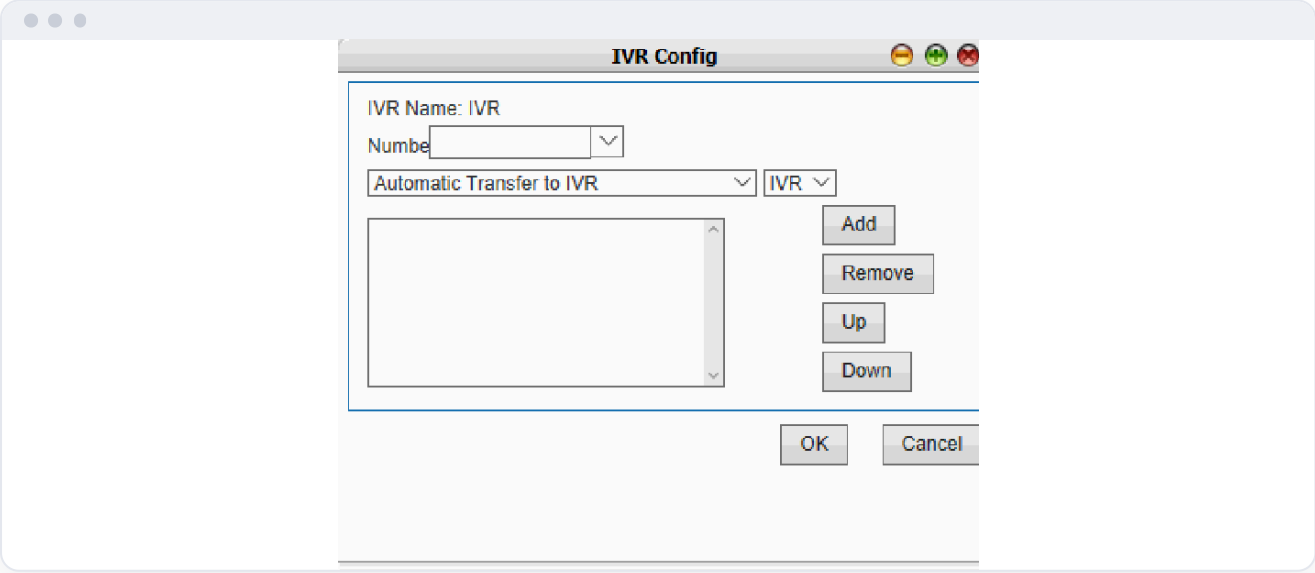


Figure 10. IVR configuration with actions per key.

Key / event	Meaning

1 (start)	Runs when the call is answered: typically play the greeting
Digits	Caller presses a key: transfer to an extension, queue, sub-menu…
2 (invalid)	Caller dialled an invalid key
0 (timeout)	Caller did nothing in the set time

Actions: transfer to IVR, extension, play prompt (interruptible or not), voicemail, group call, queue, hang up, answer and conference. Upload greetings in **PBX Settings › Prompt Tone**.

> **TIP**
> Keep menus short: at most 4–5 options, and always offer a way to reach a person. Allow direct dialling of extensions in the main menu.

SECTION 13

Queues and operator console

Queues distribute calls to a team; the soft console gives operators a live view.

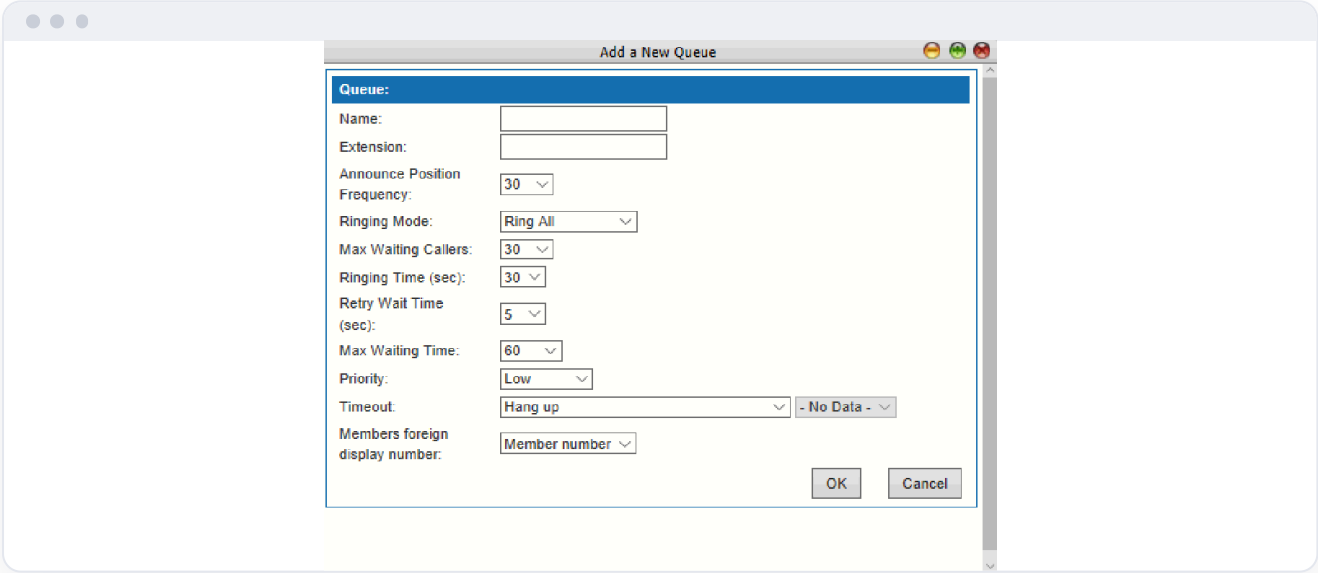


Figure 11. New queue settings.

Queue setting	Description
Name / extension number	The number callers or the IVR reach
Ringing mode	Ring all (default), round robin, least recent, fewest calls, random, round robin with memory
Announce position frequency	How often callers hear their position (default 30 s)
Max waiting callers / max waiting time	Capacity and patience limits

Ringing time / retry wait	How long an agent rings and the pause between tries
Priority	Low, medium, high or very high against other queues
Timeout action	Extension, prompt, other queue or hang up
Member display number	Show the member, queue or a custom number on outbound calls

1 Add members

After saving, click **Add** and add agents or phones as members.

SoftConsole: create operator roles with permissions, then operator accounts, for reception staff who transfer calls and monitor extensions.

> **NOTE**
> Queue numbers must not overlap with extension, conference or other service numbers.

SECTION 14

Voicemail, Follow Me and call features

Personal features that make every extension more useful.

Feature	Settings and use
Voicemail	Seconds before voicemail, up to 1000 messages, min/max length, email subject and signature. Listen with *97 / *98. Needs SMTP settings
Follow Me	Ring all, one by one or round robin with memory; ringing interval and total timeout
Secretary	Calls to a manager reach the secretary first (global, internal or external calls)
Hotline	Lifting the handset dials a fixed number (lifts, emergency phones)
Group pickup	Members answer each other's ringing phones with *115
Speed dial	Ten short codes per user (*0 + digit)
Alarm clock	Wake-up calls, for hotels
Blacklist / whitelist	Block or always allow numbers
Music ring	Ringback music: 8 kHz, 16-bit mono MP3, WAV, A-law or µ-law files

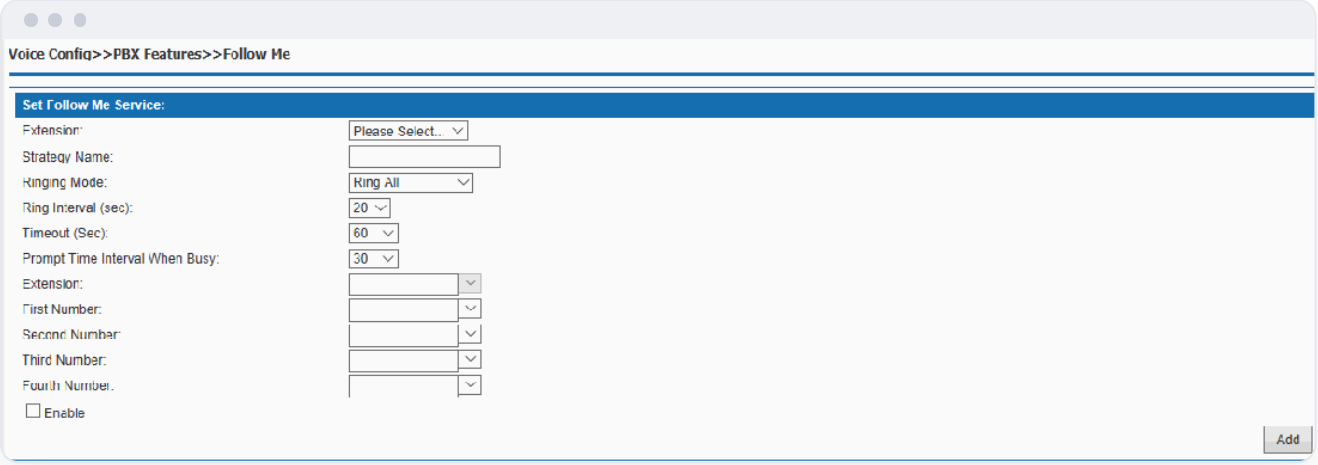


Figure 12. Follow Me strategy.

SECTION 15

Recording, conferences and billing

Record calls for quality and compliance, host conference calls and charge calls.

1 Call recording

PBX Features › Call Recording: tick **Enable Record**, enter the IP address and port of the recording server and select the extensions or departments to record. The recording server stores, plays and downloads the recordings.

> **IMPORTANT**
>
> Recording calls is regulated in most countries. Inform callers (for example with an IVR announcement) and follow local law and your data protection policy.

2 Conference bridge

PBX Features › Conference Bridge › Add: conference number, name, maximum participants and an optional PIN for protected conferences.

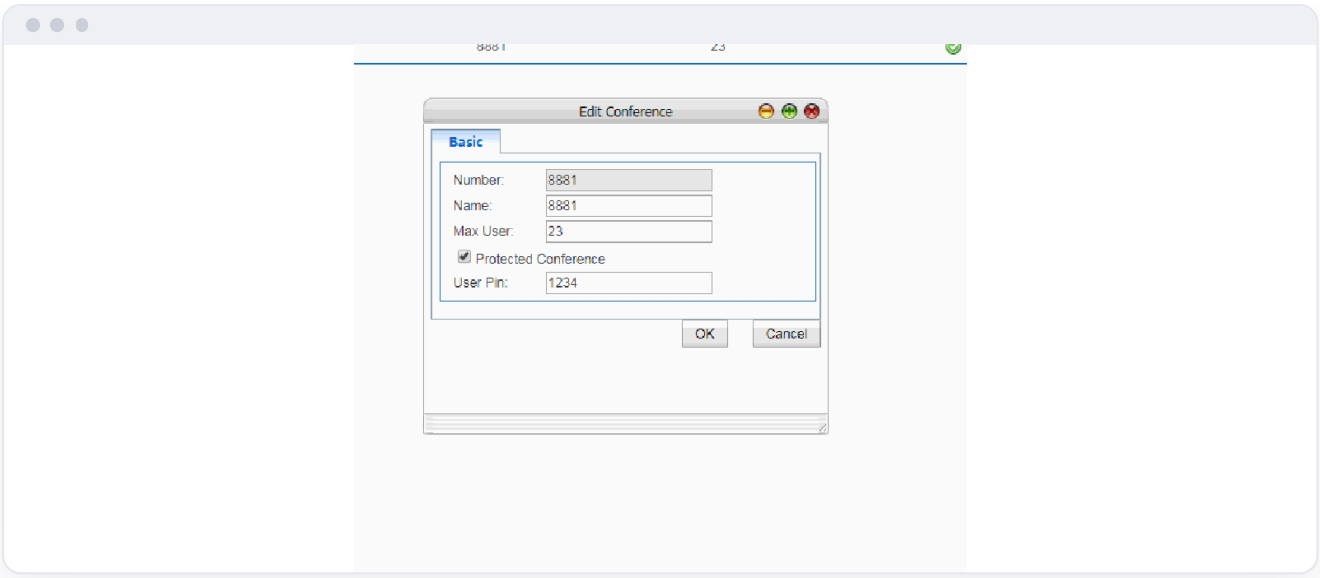


Figure 13. Conference room settings.

3 Billing

Billing Setting: connect an external billing server (interface V2.0: address, port, user and password) and set card access by call type. Prepaid and card users are set per user.

SECTION 16

Feature codes

Users control features from any phone by dialling a code. Codes can be changed in PBX Features › Feature Code.

Code	Function
`**` + ext	Pick up a call ringing on that extension
`*115`	Group pickup
`*0` + digit	Speed dial (`*75` to program)
`*72` / `*73`	Forward all calls on / off
`*90` / `*91`	Forward on busy on / off
`*52` / `*53`	Forward on no answer on / off
`*78` / `*79`	Do Not Disturb on / off
`*103` / `*104`	Absent on / off
`*70` / `*71`	Call waiting on / off
`*97` / `*98`	Voicemail: local / remote
`*30` / `*31` / `*32`	Add to blacklist / remove / blacklist last caller

`*37` / `*38` / `*59`	Call busy back on / off / register
`*105` / `*106`	Phone login / logout (hot desking)
`*111`	Hear own extension number
`*114`	Hide caller ID for the next call
`*56`	Set alarm clock (wake-up call)
`*88`	Password dialling (call with your PIN from any phone)
`*77` / `*99`	Record a prompt / play it back
`*201` / `*202` / `*203` / `*204`	Card dialling / card PIN and balance / extension balance / remaining minutes
`*33` / `*34` / `*35` + ext	Supervisor: listen / barge in / force release

IMPORTANT

Restrict the supervisor codes (*33, *34, *35) to supervisors, and inform staff where listening in is used.

TIP

Print the most used codes on a card next to each phone, or in the hotel room directory.

SECTION 17

PBX settings

System-wide voice settings.

Page	Purpose
Global Setting	System behaviour: VIP levels, secretary mode, billing options
Route Group	Groups of outbound routes assigned to users and trunks
PCM Settings	Enable E1/T1 circuits, framing (CCS/CAS) and channel mapping
Prompt Tone / Record File	Upload and manage voice prompts and recordings
VoIP Security	Lock SIP accounts and IP addresses after failed registrations
VoIP Config	SIP and RTP ports and timers
Analog Setting	FXS/FXO port parameters, gains, caller ID and batch edit
DSP Setting	Echo cancellation, jitter buffer and codec resources
SMTP Setting	Mail server for voicemail to email

License	Capacity and feature licences

SECTION 18

Security and toll-fraud protection

A PBX connected to the internet is attacked within hours. Use every layer of protection.

VoIP Security setting	Default
SIP account verification	On: 3 failed registrations in 3 minutes lock the account for 3 minutes
IP address verification	On: 3 failures in 3 minutes lock the IP for 3 minutes
Blacklist rule	An IP locked more than 3 times in 30 minutes is blacklisted
Daily long-distance limit	Warning and cut-off minutes per day (national default 3600 / 3700)

- Change the admin password, every SIP registration password (default Aa111111) and every voicemail PIN (default 0000).
- Set **Permitted registry address range** on every SIP extension to your LAN or VPN networks.
- Give international permission only where needed, and set daily call-duration limits.
- Keep web management from the WAN off (default). Use a VPN for remote administration; if needed, use HTTPS on port 443.
- Enable the firewall, ARP defence and DDoS protection in the **Security** module.
- Review the call log every week for unusual destinations or night-time calls.

> **IMPORTANT**
> Never expose the SIP port of the PBX to the internet without registration address limits and the VoIP security lock. Remote phones should connect over VPN.

SECTION 19

QoS, objects and network security

Keep voice quality high and the network protected.

- **Device QoS**: set the real WAN upload and download speeds (not set by default) so the PBX can prioritise voice. Internal VoIP packets are prioritised by default and marked with high-priority DSCP.
- **Behaviour policy**: internet access limits, a maximum number of connections (default 65536) and MAC filtering.
- **Object management**: schedules, accounts and common ports reused by policies.
- **Security**: firewall (on by default), ARP defence with gratuitous ARP, and DDoS protection.

> **TIP**
> If calls break up when someone downloads a large file, set the WAN speeds in QoS to about 90% of the real line speed.

SECTION 20

Monitoring and maintenance

Watch the system and keep it up to date.

Page	Use
Status › System / Interface	Device overview and port status
Info Statistics › Online User	Registered phones and connected users
Stat Report › Call Log	All calls, with search and export
Stat Report › Service Voice Status	Live calls and channels
Stat Report › Data capture	Packet capture for troubleshooting with Wireshark
Stat Report › Hardware status	Boards and ports
System › SNMP / TR-069	Monitoring and remote management
System › Log Manage	System logs

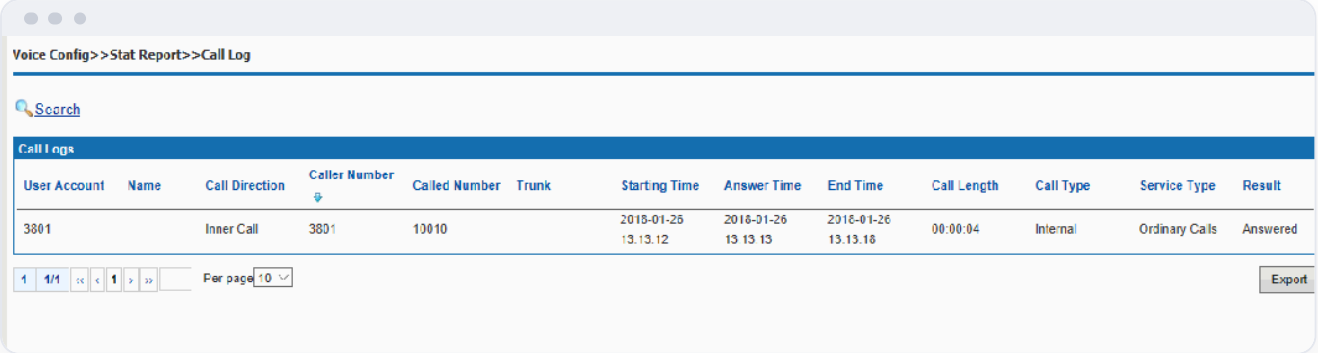


Figure 14. Call log.

1 Back up

System Management › Maintain: export the configuration after every change and store it off the PBX.

2 Upgrade

System Management › Upgrade: select the firmware file from Navijin and start the upgrade. Do it outside business hours.

- **Time Settings**: set the time zone and NTP so call logs and voicemail times are correct.
- **Reboot** and **Restore Factory Default** are in System Management. Restoring erases all settings.

SECTION 21

Commissioning checklist

Complete before handover.

Check	Done when
Earth and power	Earthed; on UPS; surge protection on external lines
Passwords	Admin, SIP and voicemail defaults all changed
Network	Static address; WAN management off; VPN for remote access; SIP ALG off on router
Extensions	All created, registered and tested; names in directory
Trunks	Registered/up; inbound and outbound test calls made on each
Routing	Day/night inbound routes; outbound fallback tested
IVR and queues	Prompts recorded; every menu path tested
Security	Registration address limits, VoIP security and call-duration limits on
Voicemail	SMTP tested; voicemail to email received
Backup	Configuration exported and stored with site documents

SECTION 22

Troubleshooting

Answers to the most common questions.

I cannot open 192.168.100.1.

Connect to a LAN port (not WAN), set the computer to DHCP or 192.168.100.x/24, and disable any browser proxy.

A SIP phone does not register.

Check the extension number, registration password and the permitted registry address range, and that the phone's IP is not locked by VoIP security.

Calls have one-way or no audio.

A NAT or SIP ALG problem. Turn off SIP ALG on the router, set the external IP on the trunk and open the RTP port range if needed.

External calls fail with a busy tone.

Check the user's permission, that an outbound route matches the dialled prefix and that the trunk is registered or up.

Incoming calls do not reach the IVR.

Check that an inbound route covers the trunk at the current day and time.

DTMF does not work in an IVR or voicemail.

Set the same DTMF mode on both sides, usually RFC 2833.

Voicemail email is not sent.

Configure and test the SMTP settings; check the user's email address.

We received a huge phone bill.

Toll fraud. Disable international permission, change all passwords, restrict registration addresses, review the call log and contact your provider.

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