



OCEANCRM · PHONE SYSTEM

Technician guide

Installing GOIP lines, desk phones and remote access, and fixing problems

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1. How it is connected

- OceanCRM runs its own phone switch on the company's OceanCRM server. Production: `oceanexpress.tech` ; testing: `oceancrm.lk` . A box or phone belongs to the server it registers to.
- Each GOIP box registers to the switch over the internet (SIP, UDP port 5060) with its own username and password. Nothing needs to be opened on the office router.
- Phones: the browser Phone window (WebRTC), desk phones and the Ocean Attendance app (SIP).
- Remote login: the GOIP dials a VPN to the server, so its web page opens from OceanCRM anywhere (**Open device**).
- Names on the switch: lines are `t<company>_gw<id>` (for example `t2_gw3`); extensions are `t<company>_<number>` (for example `t2_101`).

2. Adding a GOIP box (line)

1. In OceanCRM: **Phone system** → **Gateways** → **New pbx gateway**. Name it, kind **GOIP**, turn on **On the OceanCRM phone system**, the number of SIMs, the operator, the SIM's own number (caller ID), which number prefixes it should carry (optional) and its order. Optionally the price per minute and monthly rent.
2. Save, then click **GOIP settings** on its row. It shows the **SIP server, username** and **password**.
3. On the GOIP (Ejoin GoIP-508): **SIP Setting**: server = the OceanCRM server (`oceanexpress.tech`), port 5060, username and password as shown, one account for the whole box (trunk) or per port as shown. Click **Submit** on that page. In the per-port SIP accounts table there is a **separate Submit**; press it too.
4. **Advanced Settings** → **Port Settings**: put **Hot-line** `1000` on every port (type it on port 1, click **[Copy]**), then **Submit**. Without a hot-line the GOIP rejects incoming mobile calls: callers hear busy and the call cuts.
5. **Save and Reboot** → **Save**. **Important**: Submit only applies a setting until the next restart; **Save** writes it to the box's memory. After any change, press Save.
6. Back in OceanCRM, the Gateways list shows **Registered** in green within a minute.

3. Checking the SIMs on the GOIP

Running Status → **Port Status** shows each port's light; **Call Status** shows each call.

Light	Meaning	Action
Green	Registered OK	—
Blue	Calling	—
Red	Register failed	Check the SIM is active, has credit, is seated well, and the antenna is on. Restart the port.
Yellow	No balance	Top up the SIM.
Grey / empty	No SIM	—

Each SIM carries **one call at a time**. With one working SIM, the box cannot take a call in and make a call out together, and transfers to outside numbers fail while it is busy.

4. Remote login (open the GOIP from anywhere)

1. In OceanCRM, on the box's row click **Remote login** → **Turn on**. Note the server, user name (for example `gw3`) and password. Use the box's **own** row: its user name must match.
2. Click the **Root certificate** link and save the file `ca.crt`.
3. On the GOIP: **System Settings** → **System Update**, **File Type** = **OpenVPN certificate** (not Firmware!), choose `ca.crt`, **Submit**.
4. **Advanced Settings** → **Network Settings** → **VPN Settings**: VPN Support **OpenVPN**, Auth Mode **Password**, Server Address = the OceanCRM server (`oceanexpress.tech`), User Name and Password from step 1, Vpn Proto **UDP**. Submit.
5. **Save and Reboot**: **Save**, then **Reboot**.
6. After a minute, OceanCRM shows **Connected** and an **Open device** button. The address is `https://oceanexpress.tech/gw-web/<id>/`.

Never upload the certificate with File Type "Firmware". The box checks it and refuses ("md5 hash failed"), but a wrong firmware file could damage it.

5. Desk phones, softphones and the mobile app

1. **Phone system** → **Extensions** → **New**: number, the staff member, **Used on** = **Softphone or desk phone (SIP)**.
2. Click **Login details**: server, username (for example `t2_103`), password, transport UDP 5060.
3. Enter them into the desk phone's SIP account (registrar/proxy = server). Codec: G.711 (PCMU/PCMA).
4. The Ocean Attendance app: the staff member opens **Phone** in the app; the extension must be a SIP one.

5. A browser and a phone cannot share one extension. A person using both needs two extensions.

6. Testing a line

1. From any phone, call the SIM's number. You should hear the company greeting.
2. Press a key; the queue music plays and agents' phones ring.
3. Answer, talk 20 seconds, hang up.
4. In **Phone system** → **Call log**, the call shows the line, the wait, the talk time, the recording and the call quality (MOS).
5. Call out from a browser phone to a mobile number to test the outgoing side.

7. Fixing problems

Symptom	Likely cause	Fix
Caller hears busy, the call cuts at once	No hot-line on the GOIP port (often lost after a restart)	Port Settings: hot-line 1000, Submit, then Save .
Gateways shows "Not registered"	Wrong server, username or password; box offline; per-port SIP table not submitted	Re-enter from GOIP settings ; press both Submits; check the box's internet.
Calls come in but nobody's phone rings	No agent Ready, or the browser Phone window closed	Agents set Ready and keep the Phone window open; check the queue's members.
One-way or no audio	Firewall blocking audio (RTP), or microphone blocked	Allow the microphone in the browser; let the box reach the internet on UDP.
Calls out or transfers fail	All SIM channels busy, or no line carries that prefix	Free a SIM or add SIMs; check the gateway's prefixes.
Remote login: "TLS handshake failed"	The box does not have this server's certificate	Import <code>ca.crt</code> from this server (section 4), Save, Reboot.
Remote login connected but the page shows 404	Address without the last "/"	Use the Open device button.
Bell says "Line ... is down"	The box lost power or internet	Check power, cable/router; it re-registers by itself.
Poor call quality (MOS below 3.6)	Weak mobile signal or busy internet	Move the antenna or box; check the office internet; see Costs and call quality .

8. For OceanCRM support staff (server)

- Apply call centre configuration to a switch: `sudo bash scripts/pbx/callcentre.sh` (no restart; live calls continue).
- Rewrite every company's call flows: `sudo -u www-data php artisan pbx:rebuild`.
- Never run artisan as root on the servers; use `sudo -u www-data`.
- Logs: `/var/log/asterisk/messages`, `queue_log`, `cdr-csv/Master.csv`; the VPN: `journalctl -u openvpn-server@gw`.
- Back up the database before every production change.