

Grandstream Networks, Inc.

**Grandstream IP PBX – Account Trunk Guide**



# Introduction

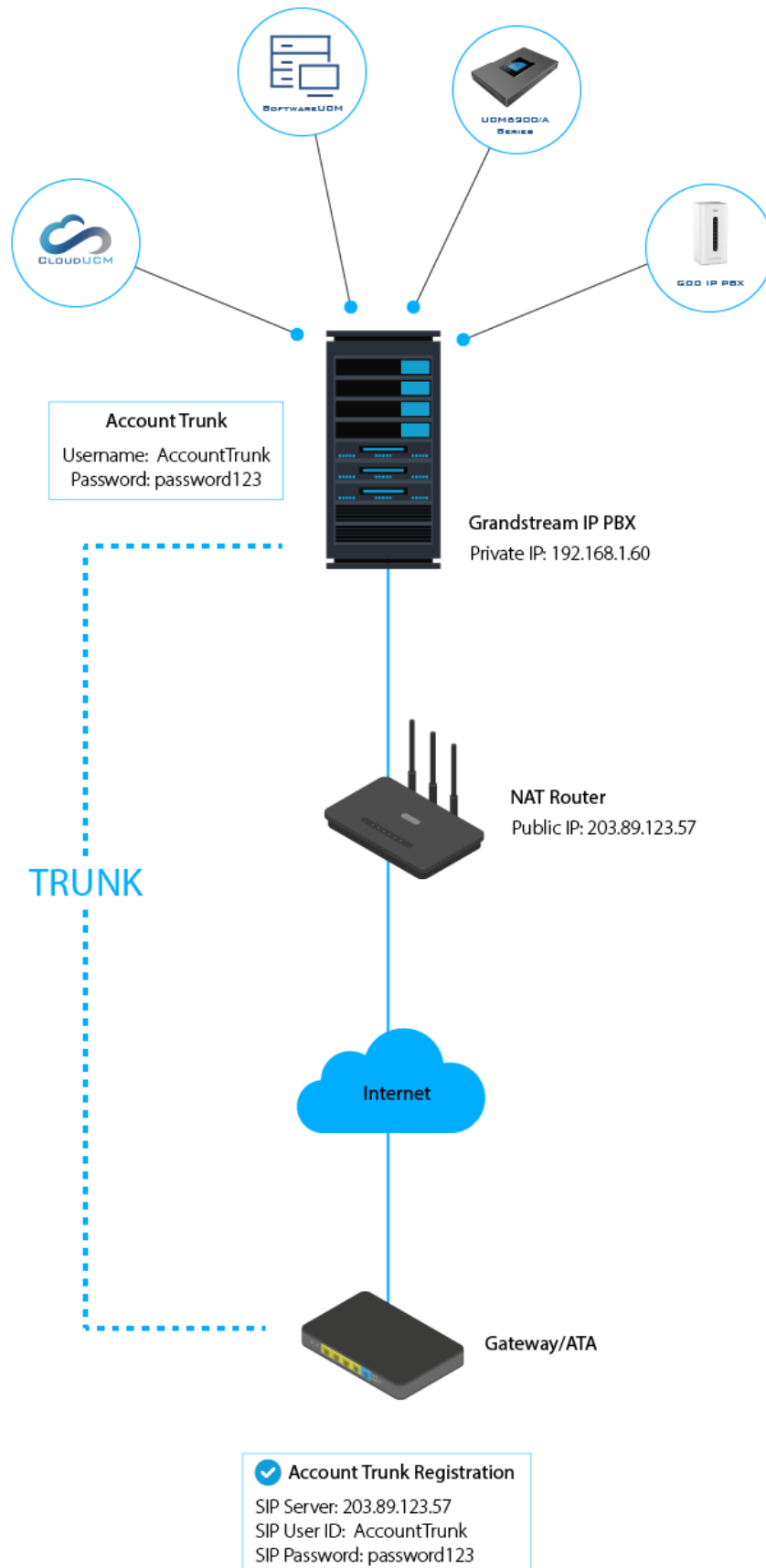
This guide is designed to help administrators configure and deploy the **Account Trunk feature** available on the Grandstream UCM6300/A series, CloudUCM, GCC IPPBX, and SoftwareUCM. This feature allows remote devices such as ATA units (HT8xx), gateway devices (HT841/HT881 series), or third-party SIP devices to register to the IP PBX as a trunk using SIP credentials, including a username and password, instead of relying exclusively on traditional IP-based peer trunking.

## Account Trunk Use Case

In traditional deployments, SIP trunking is primarily implemented through **peer-to-peer (IP-based) trunks**, which require static IP addresses, VPN tunnels, or NAT configurations. Alternatively, remote gateway devices could **register as SIP extensions**, which eliminates the ability to manage call routing as a trunk and limits the flexibility of dial plan control.

The key benefits of this implementation are:

- Remote ATA/gateway registration without static IPs.
- Centralized call routing for inbound and outbound calls.
- Improved NAT traversal and simplified firewall configuration.



# Account Trunk Configuration

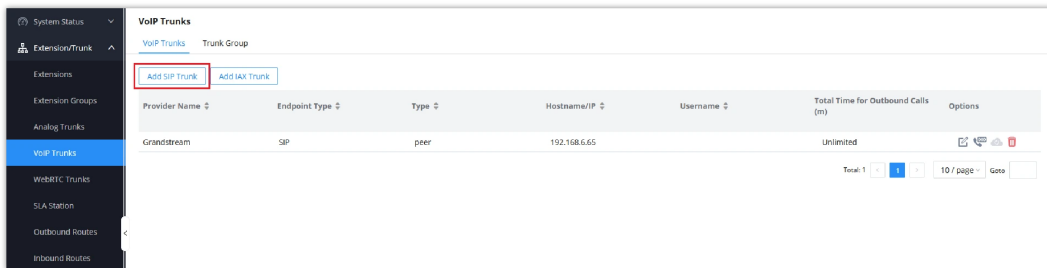
## IP PBX Configuration

This section provides a step-by-step walkthrough to configure **Account Trunking** on the Grandstream **UCM6300/A series**, **CloudUCM**, **GCC IPPBX**, and **SoftwareUCM**.

### Creating an Account Trunk

In order to create an account trunk, please refer to the steps below:

1. **Navigate** to the IP PBX Web UI.
2. Go to **Extension/Trunk** → **VoIP Trunks**, and click **Add SIP Trunk**.



3. In the **Type** drop-down, select **Account SIP Trunk**.
4. Fill in the the required fields:
  - **Provider Name**: Enter a label for the trunk (e.g., **ATA1**)
  - **Username**: The SIP account username the remote device will use to register
  - **Password**: The SIP password for authentication

**VoIP Trunks > Create New SIP Trunk**

TypeAccount SIP Trunk

\* Provider NameATA1

TransportTLS

Server Address

Keep Original CID☐

Keep Trunk CID☐

TEL URIDisabled

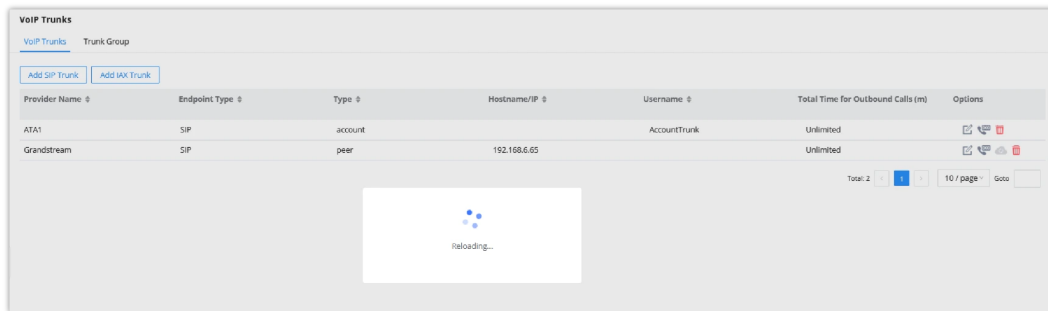
CallerID Number

CallerID Name

\* UsernameAccountTrunk

\* Password

5. Click **Save**, then **Apply Changes**.



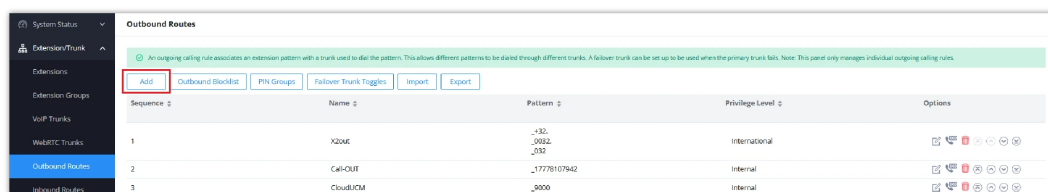
This guide focuses on the **minimum required settings**. For advanced configuration options (e.g., NAT traversal, codecs, transport types), refer to the relevant **IP PBX User Manual**.

4.

## Configuring an Outbound Route

This step defines how PBX extensions can place outbound calls via the Account Trunk by specifying the dial patterns to be routed through the trunk and determining which extensions are permitted to use it.

1. Go to **Extension/Trunk → Outbound Routes**, then click **Add**.



2. Set a **Route Name** (e.g., **To\_ATA**).

3. Define a **Dial Pattern** (e.g., **9XXXXXXXXX** to match 10-digit numbers).

4. Under **Privilege Level**, choose an appropriate level (e.g., **Internal**, **Local**, **National**, or **International**) based on which extensions should be allowed to use the route.

5. Under **Use Trunk**, select the **Account Trunk** you created earlier.

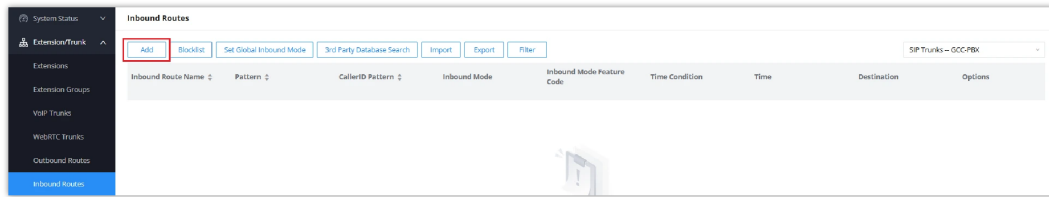
6. Click **Save**, then **Apply Changes**.

For advanced outbound route features (e.g., time conditions, failover trunk, strip/add digits), refer to the official guide: [How to Manage Inbound/Outbound Routes](#)

## Creating an Inbound Route

This section allows users to configure how the PBX handles incoming calls from the connected device (e.g., ATA, gateway, or third-party device), such as directing them to an extension, IVR, or ring group.

1. Go to **Extension/Trunk** → **Inbound Routes**, then click **Add**.



2. Under **Trunk**, select the **Account Trunk** created earlier, and choose a name for this route.
3. Enter a **DID Pattern** (e.g., `_x.` to match all numbers).

4. Set a **Default Destination** (e.g., an Extension, IVR, or Ring Group).

5. Click **Save**, then **Apply Changes**.

For more advanced inbound routing options (e.g., DID Mode, caller ID filtering, time conditions), refer to the official guide: [How to Manage Inbound/Outbound Routes](#)

## Configuring the Gateway/ATA Device

To establish a connection between the Grandstream IP PBX system and an analog gateway or ATA using the **Account Trunk** feature, you will need to configure the gateway/ATA to register as a SIP trunk using the credentials generated on the IP PBX.

This configuration applies to any SIP-compatible analog gateway or ATA device that supports SIP account registration, including FXO/FXS gateways. In this guide, we will be using the **Grandstream HT818 ATA** as an example device.

### Example: Configuring the Grandstream HT818 ATA

In this example, we'll walk through how to configure the **Grandstream HT818 ATA** to register FXS ports to the Grandstream IP PBX using the **Account Trunk** method. This allows analog phones or analog line interfaces connected to the HT818 to be treated as trunk endpoints or routed extensions via SIP registration.

## Configuring SIP Server under Profile Settings

Go to **HT818 Web UI** → **Profile 1 (or Profile 2)**, and configure the following:

- **Primary SIP Server:** Enter the IP address or domain name of the IP PBX
- **SIP Transport:** Match the SIP transport settings of the IP PBX
- **NAT Traversal:** Configure based on network requirements
- **SIP Registration:** Set to “Yes”
- **Register Expiration:** 60 minutes is typical

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS PROFILE 1 PROFILE 2 FXS PORTS

Profile Active: ☐ No ☒ Yes

Primary SIP Server:  (e.g., sip.mycompany.com, or IP address)

Failover SIP Server:  (Optional, used when primary server no response)

Prefer Primary SIP Server: ☒ No ☐ Will register to Primary Server if Failover registration expires ☐ Will register to Primary Server if Primary Server responds, need to enable SIP OPTIONS/NOTIFY Keep Alive

Outbound Proxy:  (e.g., proxy.myprovider.com, or IP address, if any)

Backup Outbound Proxy:  (e.g., proxy.myprovider.com, or IP address, if any)

Prefer Primary Outbound Proxy: ☒ No ☐ Yes (yes - will reregister via Primary Outbound Proxy if registration expires)

From Domain:  (Optional, actual domain name, will override the from header)

Allow DHCP Option 120 (override SIP server): ☒ No ☐ Yes

SIP Transport: ☐ UDP ☐ TCP ☒ TLS (default is UDP)

SIP URI Scheme When Using TLS: ☐ sip ☒ sips

Use Actual Ephemeral Port in Contact with TCP/TLS: ☒ No ☐ Yes

NAT Traversal: ☐ No ☐ Keep-Alive ☐ STUN ☐ UPnP ☒ Auto ☐ VPN

For more advanced SIP parameters (e.g., Outbound Proxy, DNS mode), refer to the [HT818 Administration Guide](#).

## Configuring FXS Port Registration

Navigate to **HT818 Web UI** → **FXS Ports**, choose the port to use, and fill in the information below:

- **SIP User ID:** Enter the username from the IP PBX Account Trunk
- **Authenticate ID:** (if applicable) Enter the AuthID from the IP PBX Account Trunk
- **Name:** (Optional display name)
- **Profile:** Select the profile configured in the first section (e.g., *Profile 1*)
- **Enable Port:** Set to “Yes”

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS PROFILE 1 PROFILE 2 FXS PORTS

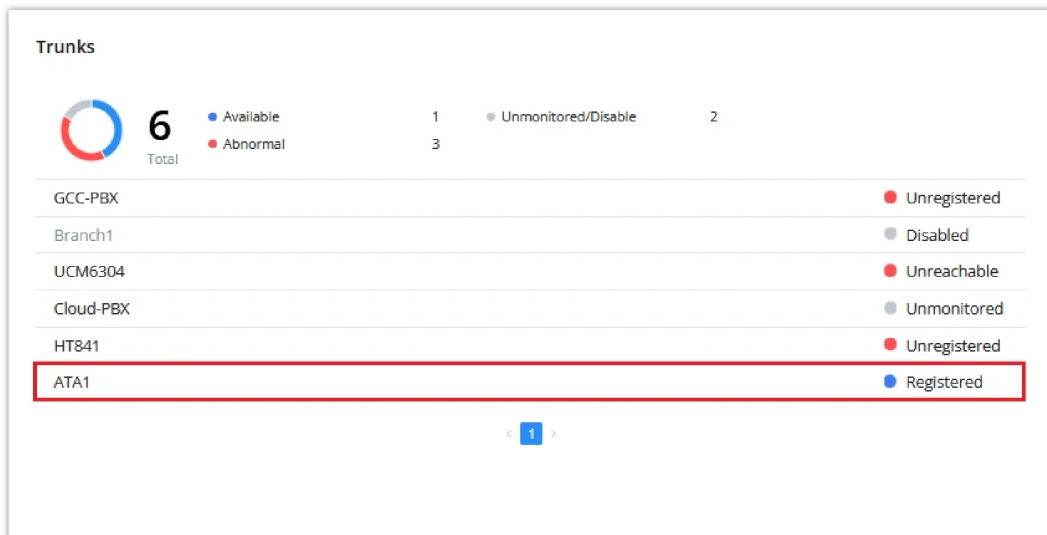
User Settings

| Port | SIP User ID  | Authenticate ID | Password | Name | Profile ID  | Enable Port                                                   |
|------|--------------|-----------------|----------|------|-------------|---------------------------------------------------------------|
| 1    | AccountTrunk |                 | .....    |      | Profile 1 v | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| 2    |              |                 |          |      | Profile 1 v | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| 3    |              |                 |          |      | Profile 1 v | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| 4    |              |                 |          |      | Profile 1 v | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| 5    |              |                 |          |      | Profile 1 v | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| 6    |              |                 |          |      | Profile 1 v | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| 7    |              |                 |          |      | Profile 1 v | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| 8    |              |                 |          |      | Profile 1 v | <input type="radio"/> No <input checked="" type="radio"/> Yes |

## Verifying Registration and Testing

## 1. Verify Account Trunk Registration

Log in to the IP PBX web UI and navigate to the **Dashboard** view to confirm the **Account Trunk** status shows as "**Registered**".



## 2. Verify Internal Extension Registration

Ensure your internal SIP extensions (e.g., desk phones or softphones) are registered and ready to make calls.

## 3. Test Calling from Analog to SIP Extension

- Pick up the analog phone connected to the HT818's FXS port.
- Dial the number of a SIP extension registered to the IP PBX (e.g., 1001).
- Verify that the SIP extension rings, and you can establish two-way audio.



## 4. Test Calling from SIP Extension to Analog Phone

- On the IP PBX, navigate to **Extension/Trunk** → **Outbound Routes**, and locate the outbound rule associated with the Account Trunk.
- Add a dial pattern for reaching the analog phone (e.g., 123456).
- From a SIP extension, dial 123456.
- The IP PBX will route the call through the Account Trunk to the HT818, and the analog phone should ring.
- Answer the call and verify two-way audio.

Outbound Routes > Edit Outbound Rule: To\_ATA

General

\* Outbound Rule Name

To\_ATA

\* Pattern

\_X.

\_123456

PIN Groups

None

Password

Local Country Code

A black Grandstream UCM6300/A Series IP phone is shown from a front-three-quarter view. The phone has a large color LCD screen at the top displaying a call interface with a blue header, a green status bar, and a numeric keypad. Below the screen is a circular navigation pad and several function buttons. The main keypad is a standard 12-button numeric keypad. The phone is on a black base with a handset on the left side.

## Supported Devices

| Model            | Firmware Support |
|------------------|------------------|
| UCM6300/A Series | 1.0.27.10+       |
| CloudUCM         | 1.0.25.18+       |
| GCC IPPBX        | 1.0.27.14+       |
| SoftwareUCM      | 1.0.27.18+       |