

Manage Analog Lines

UCM6XXX series have integrated FXO ports allowing connection of PSTN lines and/or lines from traditional analog PBX. An analog trunk can be defined as a single line or multiple lines (PSTN lines or lines from traditional analog PBX). Each UCM6XXX supports at least 2 FXO ports and reaching 8 FXO ports for UCM6208 and UCM6308/A models.

Using analog trunks allow customers to keep using their actual PSTN lines/PBX lines and integrating them to VoIP systems such as UCM6xxx giving it hybrid capability from VoIP to PSTN and vice versa and taking VoIP and other services offered by UCM6xxx including mobility and reducing calling costs, call recording, IVR, privileges, and rules for users, ability to use analog trunk from remote extensions...

In this guide, we will be using UCM6302 as an example to provide steps on how to configure Analog Trunks as well as basic Inbound/Outbound Rules, ACIM Settings, and PSTN Detection configuration, and how to monitor analog trunks using SLA Station.

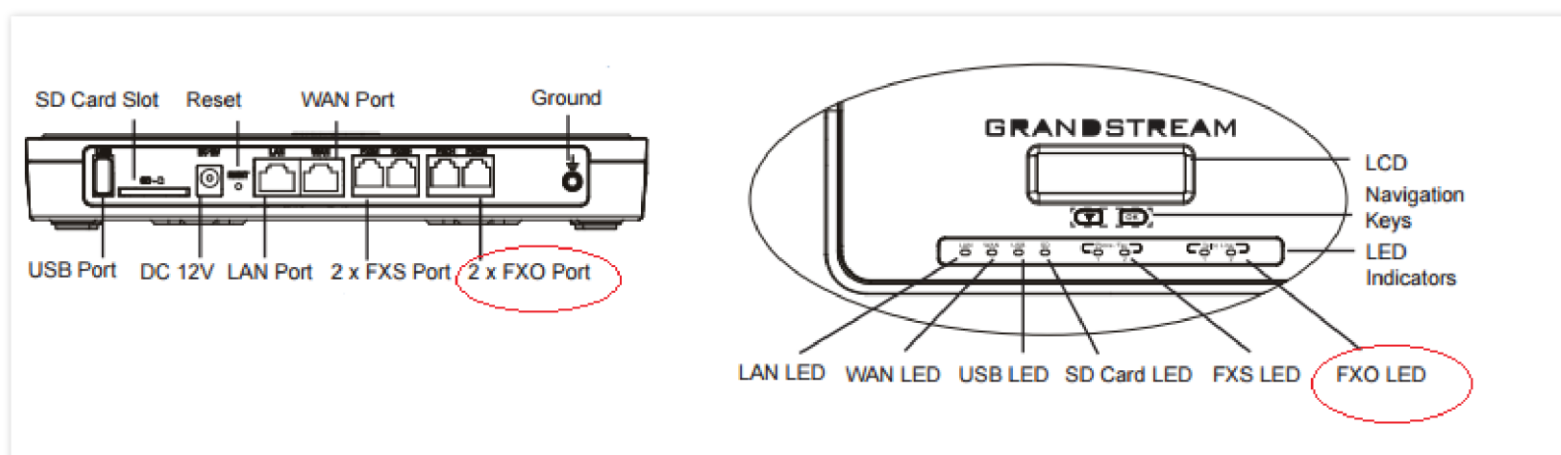


- UCM6xxx Series include UCM6200 Series, UCM6510 and UCM630x/A series.
- UCM6200 series supports 2/4/8 ports.
- UCM6510 supports 2 FXO ports.
- UCM6300 series supports 1/2/4/8 ports.
- UCM6300A series supports 2/4/8 ports.

UCM6XXX Setup Overview

To connect analog lines to UCM6xxx, please follow the below steps:

- Connect the UCM6XXX to the Network via WAN/LAN Port, and power it on.
- Once the UCM6XXX boots up and connects to the network, the corresponding LAN/WAN LED indicators will be in solid green and the LCD shows the IP address
- Connect the Analog lines from the wall jack/CO switch to the FXO ports, the corresponding FXO LEDs will be in solid green.



UCM6202 Front and Back Panels

Once the FXO ports are physically connected to analog lines, users may log in to the UCM6XXX's Web GUI to configure the Analog Trunks settings and outbound/inbound rules.

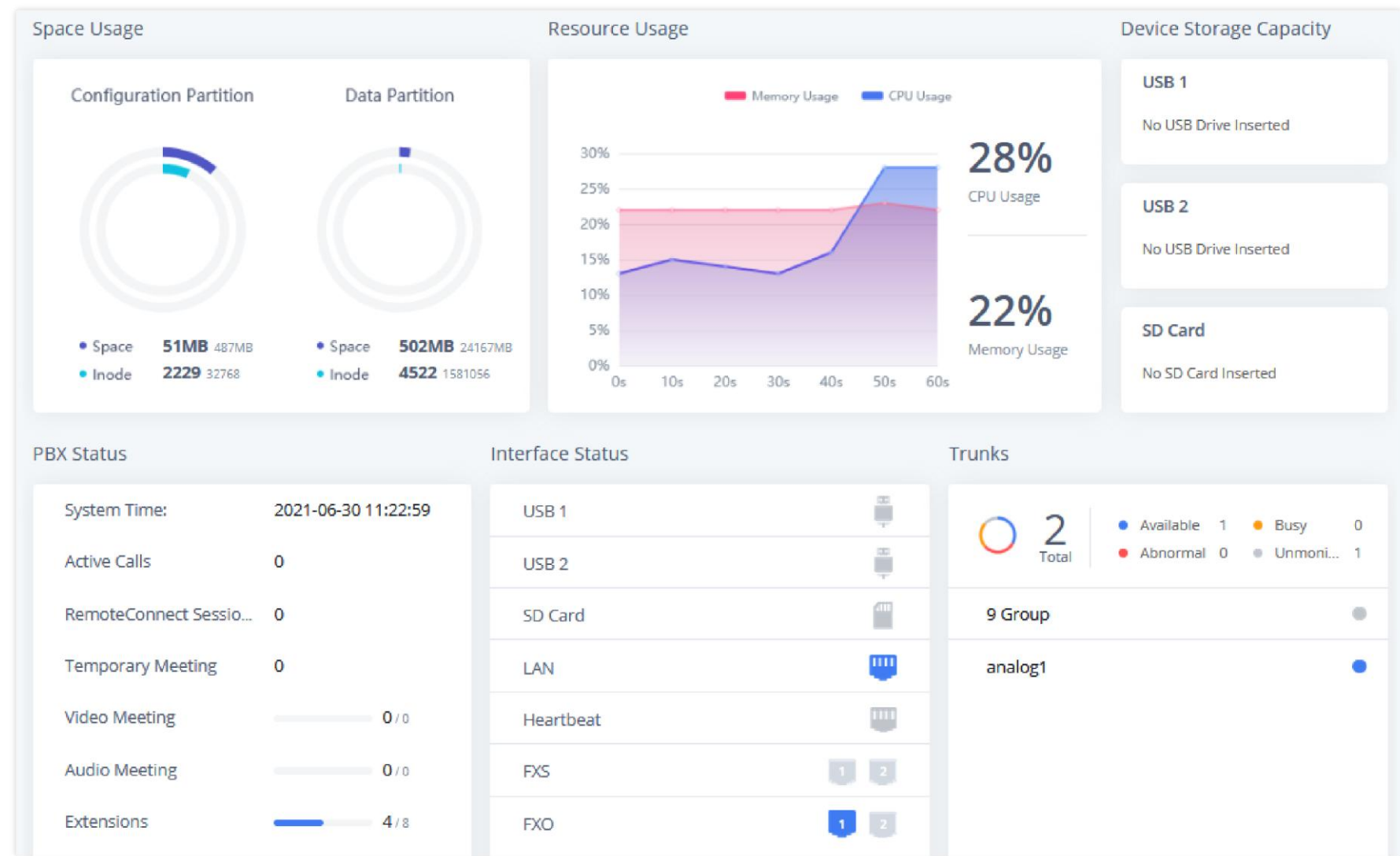
Configuring Analog Trunks

Use a computer connected to the same network as the UCM6XXX, then login to the web GUI of the UCM6XXX by typing the IP address shown on its LCD on your browser, and type in the admin username and password.

PBX Status

After login, the first default page will show the UCM6XXX status, this will help users to check the FXO Ports status and give the ability to quickly see the FXO overview, without physically monitoring the FXO LEDs.

When connecting a new analog line to an FXO port, its interface status on the web GUI will be in solid grey and shows *connected but not configured*, as shown in the below figure.



FXO Port Connected but not Configured

FXO icon in green indicates analog trunk is ready, while in orange color, it indicates the trunk is used.

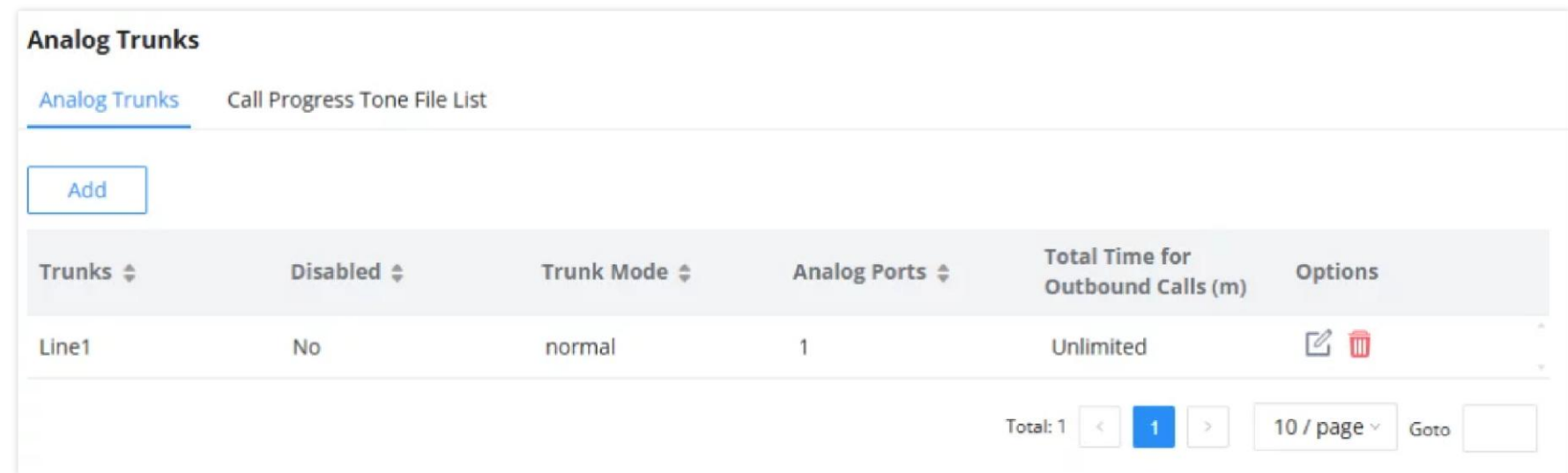
Configure a New Analog Trunk

This section will give an overview of how to create/edit analog trunks, and detect PSTN parameters.

Create a New Analog Trunk

To create a new analog trunk:

- 1. Go to Web GUI→PBX→Extension/Trunk→Analog Trunks.
- 2. Click on + Add to add a new analog trunk.



Create a New Analog Trunk

- 3. The following figure will pop up, allowing users to configure the connected analog lines to the FXO port.

* FXO Port

☒ 1
☐ 2

* Trunk Name

Line1

Advanced Options

Disable This Trunk

☐

SLA Mode

☐

Enable Polarity Reversal

☐

Current Disconnect Threshold (ms)

☒

200

* RX Gain

0dB

Use CallerID

☒

Caller ID Scheme

Bellcore/Telcordia

Auto Record

☐

Append #

☐

DAHDI Out Line Selection

Ascend

Echo Cancellation Mode

Default

Noise Cancelation RX Frequency (Hz)

OFF

* Ring Timeout

8000

* TX Gain

0dB

Fax Mode

None

* FXO Dial Delay (ms)

0

Direct Callback

☐

* Max Outgoing Calls

0

* Max Incoming Calls

0

Noise Cancelation TX Frequency (Hz)

OFF

Cancel

Save

Analog Trunk Configuration Page

Please refer to the following table for Analog Trunk configuration parameters.

Parameter	Description
FXO Port	Select the channel for the analog trunk. • UCM6301: 1 channel • UCM6302: 2 channels • UCM6304: 4 channels • UCM6308: 8 channels
Trunk Name	Specify a unique label to identify the trunk when listed in outbound rules, incoming rules, etc.
Advanced Options	
Disable This Trunk	If selected, the trunk will be disabled, and incoming/Outgoing calls via this trunk will not be possible.
SLA Mode	Enable this option to satisfy two primary use cases, which include emulating a simple key system and creating shared extensions on a PBX. Enable SLA Mode will disable polarity reversal.
Barge Allowed	The barge option specifies whether other stations can join a call in progress on this trunk. If enabled, the other stations can press the line button to join the call. The default setting is Yes. This option appears when SLA Mode is enabled.

Hold Access	The hold option specifies hold permissions for this trunk. If set to “Open”, any station can place this trunk on hold and any other station is allowed to retrieve the call. If set to “Private”, only the station that places the call on hold can retrieve the call. The default setting is Yes.
Enable Polarity Reversal	If enabled, a polarity reversal will be marked as received when an outgoing call is answered by the remote party. For some countries, a polarity reversal is used for signaling the disconnection of a phone line and the call will be considered as “Hangup” on a polarity reversal. The default setting is “No”.
Polarity on Answer Delay (ms)	When the FXO port answers the call, FXS may send a Polarity Reversal. If this interval is shorter than the value of “Polarity on Answer Delay”, the Polarity Reversal will be ignored. Otherwise, the FXO will On-hook to disconnect the call. The default setting is 600ms.
Current Disconnect Threshold (ms)	This is the periodic time (in ms) that the UCM630X will use to check on a voltage drop in the line. The default setting is 200. The valid range is 50 to 3000.
Ring Timeout (ms)	Configure the ring timeout (in ms). Trunk (FXO) devices must have a timeout to determine if there was a Hangup before the line is answered. This value can be used to configure how long it takes before the UCM630X considers a non-ringing line with Hangup activity. The default setting is 8000.
RX Gain	Configure the RX gain for the receiving channel of the analog FXO port. The valid range is from -13.5 (dB) to + 12.0 (dB). The default setting is 0.
TX Gain	Configure the TX gain for the transmitting channel of the analog FXO port. The valid range is from -13.5 (dB) to + 12.0 (dB). The default setting is 0.
Use CallerID	Configure to enable CallerID detection. The default setting is “Yes”.
Caller ID Scheme	Select the Caller ID scheme for this trunk. • Bellcore/Telcordia • ETSI-FSK During Ringing • ETSI-FSK Prior to Ringing with DTAS • ETSI-FSK Prior to Ringing with LR • ETSI-FSK Prior to Ringing with RP • ETSI-DTMF During Ringing • ETSI-DTMF Prior to Ringing with DTAS • ETSI-DTMF Prior to Ringing with LR • ETSI-DTMF Prior to Ringing with RP • SIN 227-BT • NTT Japan • Auto-Detect If you are not sure which scheme to choose, please select “Auto Detect”. The default setting is “Bellcore/Telcordia”.
Fax Mode	Configures how faxes to this extension will be handled. • None: Faxes will not be processed. • Fax Gateway: Faxes to this extension will be processed and converted from T.30 to T.38 or vice-versa. FXS/FXO ports only. The default setting is None.
FXO Dial Delay (ms)	Configure the time interval between off-hook and first dialed digit for outbound calls.
Auto Record	Enable automatic recording for the calls using this trunk. The default setting is disabled. The recording files can be accessed under Web GUI > CDR > Recording

	Files.
Speed Dial Out	If enabled, a # sign will be automatically added to the end of the called number for outgoing calls to improve dialing efficiency. If the FXS gateway does not support #, it may cause outgoing calls to fail. After activation, the SLA mode will be disabled.
Direct Callback	Allows external numbers the option to get directed to the extension that last called them. For Example, User 2002 has dialed external number 061234575 but they were not reachable, once they have received missed call notification on their phone, they would mostly call back the number, if the option “Direct Callback” is enabled then they will be directly bridged to user 2002 regardless of the configured inbound destination.
DAHDI Out Line Selection	This is to implement an analog trunk outbound line selection strategy. Three options are available: ● Ascend: When the call goes out from this analog trunk, it will always try to use the first idle FXO port. The port order that the call will use to go out if UCM6302 is used would be port 1→port 2→. Every time it will start with port 1 (if it is idle). ● Poll: When the call goes out from this analog trunk, it will use the port that is not used last time. And it will always use the port in the order of port 1→2→1→2→1→2→..., following the last port being used in case UCM6302, is used. ● Descend: When the call goes out from this analog trunk, it will always try to use the last idle FXO port. The port order that the call will use to go out if UCM6302 is used would be port 2→port 1. Every time it will start with port 2 (if it is idle). The default setting is “Ascend” mode.
Max Outgoing Calls	The number of current outgoing calls over the trunk at the same time. The default value 0 means no limit.
Echo Cancellation Mode	The Non-Linear Processing (NLP) in echo cancellation helps to remove/suppress residual echo components that could not be removed by the LEC (Line Echo Canceller). Following modes are supported: ● Default: The NLP limits the signal level to the background noise level when active, and the background noise level adjustment is low. ● High Noise Level Adjustment: The NLP limits the signal level to the background noise level when active, and the background noise level adjustment is high. ● Noise Masking: The NLP sends sign noise when active, and the background noise level adjustment is high. ● White Noise Injection: The NLP injects white noise when active. The level corresponds to the background noise level at Sin, and the background noise level adjustment is high.
Max Incoming Calls	The max allowed number of concurrent incoming calls through the trunk. Default is 0 (no limit).
Noise Cancellation RX Frequency (Hz)	Prevents interference from low-frequency signals; when enabled, the RX frequency will only allow signals above this configured value to pass. These are the supported values: ● OFF ● 20 ● 40 ● 60 ● 200 The default value is "OFF".
Noise Cancellation TX Frequency (Hz)	Prevents interference from low-frequency signals; when enabled, the TX frequency will only allow signals above this configured value to pass. These are the supported values:

	• OFF • 20 • 40 • 60 • 200 The default value is "OFF".
Enable Total Time Limit For Outbound Calls	If enabled, a limit will be placed on the cumulative duration of outbound calls within a specified period. Once this limit has been reached, further outbound calls from this trunk will not be allowed.
Period	Configure the cumulative period for the total duration of outgoing calls. After reaching the next cycle, the accumulated time will be cleared, and it will start to accumulate again from 0. At the same time, the limit of the time-out trunk will be automatically lifted. If By Month is selected, the cycle will be one natural month, if Quarterly is selected, the cycle will be three months from the current month.
Total Time (m)	Configure the maximum allowed cumulative duration of outbound calls allowed by this trunk within the configured period. Once this limit has been reached, further outbound calls from this trunk will not be allowed.
Tone Settings	
Busy Detection	Busy Detection is used to detect far-end Hangup or for detecting busy signal. The default setting is “Yes”.
Busy Tone Count	If “Busy Detection” is enabled, users can specify the number of busy tones to be played before hanging up. The default setting is 2. Better results might be achieved if set to 4, 6, or even 8. Please note that the higher the number is, the more time is needed to Hangup the channel. However, this might lower the probability to get a random Hangup.
Congestion Detection	Congestion detection is used to detect far-end congestion signal. The default setting is “Yes”.
Congestion Count	If “Congestion Detection” is enabled, users can specify the number of congestion tones to wait for. The default setting is 2.
Tone Country	Select the country for tone settings. If “Custom” is selected, users could manually configure the values for Busy Tone and Congestion Tone. The default setting is “United States of America (USA)”.
Busy Tone	Syntax: f1=val[@level][,f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]]; Frequencies are in Hz and cadence on and off are in ms. Frequencies Range: [0, 4000) Busy Level Range: (-300, 0) Cadence Range: [0, 16383]. Select Tone Country “Custom” to manually configure Busy Tone value. Default value: f1=480@-50,f2=620@-50,c=500/500
Congestion Tone	Syntax: f1=val[@level][,f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]]; Frequencies are in Hz and cadence on and off are in ms. Frequencies Range: [0, 4000) Busy Level Range: (-300, 0) Cadence Range: [0, 16383]. Select Tone Country “Custom” to manually configure Busy Tone value.

	Default value: f1=480@-50,f2=620@-50,c=250/250
PSTN Detection	Click on “Detect” to detect the busy tone, Polarity Reversal, and Current Disconnect by PSTN. Before the detection, please make sure there is more than one channel configured and working properly. If the detection has a busy tone, the “Tone Country” option will be set as “Custom”.

Once the analog trunk is created, it can be edited anytime by clicking on  icon or deleted by clicking on  icon.

Analog Trunks

Analog Trunks

Call Progress Tone File List

Add

Trunks	Disabled	Trunk Mode	Analog Ports	Total Time for Outbound Calls (m)	Options
Line1	No	normal	1	Unlimited	 

Total: 1

<1>

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Goto

Analog Trunk Created

Create Outbound/Inbound Rules

In this section, we will give an example of creating Outbound and Inbound Rules for connected analog lines to FXO ports, to allow making and receiving calls via Analog Trunks.

Please follow the steps mention in the following section to configure the inbound and outbound routes.

 **Note**

For more details about inbound and outbound routes configuration, please refer to the following guide:
<https://documentation.grandstream.com/knowledge-base/how-to-manage-inbound-outbound-routes/>

Create Inbound Routes

- Go to **Extension / Trunk→Inbound Routes**.
- Click on “**Add**”, the following figure will appear.
- Select Analog Trunk created and Default destination (IVR, extension, ring group...).

Inbound Routes > Create New Inbound Rule

General

* Trunks

SIP Trunks -- Grandstream

* Inbound Route Name

Disable This Route
☐

Pattern

* Pattern

CallerID Pattern

CID Source

None

Seamless Transfer Allowlist

Dial Trunk
☐

Call Setting

Ringback Tone

None

Alert-info

None

Auto Record
☐

Fax Detection
☐

Block Collect Calls
☐

Cancel

Save

Create New Inbound Rule

Once the inbound rule is created, it can be edited anytime by clicking on  icon or deleted by clicking on  icon.

Default Mode
Mode 1

* Default Destination

By DID

Strip

0

Prepend

Time Condition

Add

Time Condition	Time	Week	Month	Day	Destination	Options
Office Time	Default	Default	Default	Default	IVR -- test	     

Inbound Rules with Time condition

In the above example, incoming calls during office hours will be routed to IVR, otherwise, calls will be routed to extension 1000 (Out Of Office hours).

Inbound Routes

AddBlocklistSet Global Inbound Mode3rd Party Database SearchImportExportFilter

SIP Trunks -- UCM_B

Inbound Route Name	Pattern	CallerID Pattern	Inbound Mode	Inbound Mode Feature Code	Time Condition	Time	Destination	Options
inboundd	+X. _X.	No Limit	Default Mode	-- --	-- --	Default Default	Default Mode B y DID -- Strip 0 Mode 1 User Ex tensions -- 2020	  

Total: 1

<1>

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Goto

Create Outbound Routes

- 1. Go to **Extension / Trunk→Outbound Routes**.
- 2. Click on “**Add**”, the following figure will appear.
- 3. Enter the **Calling Rule Name, Pattern** and choose the Analog trunk in **Use Trunk**.

Outbound Routes > Create New Outbound Rule

General

Outbound Rule Name

Analog_Trunk

Disable This Route

☐

Pattern

_9.

Privilege Level

National

PIN Groups

None

PIN Groups with Privilege Level

☐

Password

Auto Record

☐

Local Country Code

Enable Source Caller ID Allowlist

Enable Source Caller ID Allowlist

☐

Outbound Route CID

Call Duration Limit

Call Duration Limit

☐

Main Trunk

Cancel

Save

Create a New Outbound Rule

In this example, dialed numbers matching pattern _9X. will go through this analog trunk, if extension making the call has “National” privilege or higher. UCM6xxx will strip the leading first digit from the entered number before dialing out.

For example, to dial number 06123456789, users need to have the privilege “National” or higher on their extension(s), and dial 906123456789, UCM6xxx will receive the calling request, and since it’s matching configured pattern _9X. it will strip the leading digit (in this case 9) and dials the rest of the number (in this case 06123456789).

Outbound Routes

An outgoing calling rule associates an extension pattern with a trunk used to dial the pattern. This allows different patterns to be dialed through different trunks. For example, "local" allows 7-digit dialed through FXO port while "long distance" allows 10-digit dialed through a low-cost SIP trunk. A failover trunk can be set up to be used when the primary trunk fails. Note: This panel only manages individual outgoing calling rules.

Add

Scheduled Sync

Outbound Blocklist

PIN Groups

Failover Trunk Toggles

Import

Export

Sequence	Name	Pattern	Privilege Level	Options
1	Analog_Trunk	_9.	Internal	

Total: 1

<

1

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Outbound Rule Created

Once the outbound rule is created, it can be edited anytime by clicking on  icon or deleted by clicking on  icon.

Using Analog Trunk as Failover

Failover Trunks are used when the primary trunk fails in order to assure the continuity of external calling service for extensions.

Users can set up their Analog Trunk as a failover for another trunk by following the below steps:

1. Go to **Extension / Trunk→Outbound Routes**.
2. Click on  next to your Primary Trunk.
3. Scroll down to **Failover Trunk**, and click on **Add** to add a failover trunk.
4. Users can select then the Trunk to be set as failover as shown in figure below.

Main Trunk

* Trunk:

AnalogTrunks -- analog1

Strip:

1

Prepend:

Failover Trunk

* Trunk:

SIPTrunks -- analog2

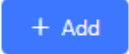
Strip:

Prepend:

Save

Cancel

Add Analog Trunk as Failover

- Click on  to add another failover trunk.
- Click on     to arrange the order. If there are multiple trunks set as failovers.
- Click on  to edit, or  to delete the Trunk from Failover list.

Main Trunk

* Trunk:

AnalogTrunks -- analog1

Strip:

1

Prepend:

Failover Trunk

+ Add

TRUNKS	STRIP	PREPEND	OPTIONS
SIP Trunks -- analog2	0		     

Analog Trunk added as Failover

ACIM And PSTN Detection

UCM6xxx series support ACIM and PSTN detections modules allowing auto-detection and configuration of PSTN lines, this helps to get the best parameters to use on PSTN lines avoiding common issues when it comes to using PSTN lines such as echo, noise, calls not ended when the remote party ends the call.

ACIM Settings

Users may sometimes face audio quality issues like noise, echo, audio volume level and choppy voice issues, this is mostly related to impedance parameters not configured correctly on the FXO ports.

Please follow the below steps to configure ACIM Settings:

1. Go to **PBX Settings→Interface Settings→Analog Hardware**.

Interface Settings

Analog Hardware

DAHDI Settings

Cancel

Save

Type

TYPE	PORT	OPTIONS
FXS Ports	1,2	
FXO Ports	1,2	

Basic Settings

* Tone Region:

United States of America (USA)

Advanced Settings

* FXO Opermode:

United States of America (USA)

* FXS Opermode:

United States of America (USA)

FXS TISS Override:

☐

* PCMA Override:

PCMU

* Boost Ringer:

Normal (40V)

* Fast Ringer:

Normal

* Low Power:

Normal

* Ring Detect:

Standard

* FXS MWI Mode:

FSK

* FXO Frequency Tolerance:

63 Hz

Ports Config Page

2. Users can select their country under **FXO Opermode** to set the On Hook Speed, Ringer Impedance, Ringer Threshold, Current Limiting, TIP/RING voltage adjustment, Minimum Operational Loop Current, and AC Impedance as predefined for your country’s analog line characteristics.

Note that different countries often have different standards for impedance. Impedance can also depend on PSTN provider/CO switches and used wires in the same country.

3. Click on under Options next to FXO Ports.
4. Users can manually enter the ACIM settings by selecting the value from dropdown list for each port. Or users could choose the detect option (either ERL or Pr) then click on “**Detect**” to automatically detect the ACIM value. The detecting value will be automatically filled into the settings.

FXO Port

ACIM Settings

FXO Signaling Type Settings

ACIM Detection

Detect

Detect Option

ERL

Port 1

600 Ω

Port 2

600 Ω

Cancel

Update

ACIM Settings

5. Click “**Update**” to apply the detected ACIM value.

After setting the AC impedance for the connected line to the FXO Port, and notice that audio gains are still low. Users can increase manually Rx/Tx gains for the **Receiving/Transmitting channel** of the analog FXO port. The Rx/Tx audio gains settings can be configured by editing the analog trunk from **PBX→Basic/Call Routes→Analog Trunks**. The valid range is from -13.5 (dB) to +12.0 (dB). The default setting is 0.

PSTN Detection

PSTN Detection module can be used to detect and configure the following parameters:

- **Busy Tone:** Depending on PSTN providers/countries/analog PBX, a busy tone signal can be used to indicate to other party (UCM6xxx) that the call has ended.
- **Polarity Reversal:** Depending on PSTN providers/countries/analog PBX, a polarity reversal signal can be used to indicate to other party (UCM6xxx) that the call has ended.
- **Current Disconnect** which is an interruption of line power from the CO switch/PSTN Provider or analog PBX to the FXO port, which last for a specific duration depending on your PSTN provider or Co Switch/PBX

Automatic Detection of PSTN Parameters

The UCM6XXX provides a PSTN detection function to help users detect the busy tone, Polarity Reversal, and Current Disconnect by making a call from the PSTN line to another destination. The detecting call will be answered and up for about 1 minute. Once finished, results will show and can be used for UCM6xxx settings.

1. Edit the analog trunk previously created.
2. In “Edit analog trunk” window, go to “**Tone Settings**” section.

Tone Country includes a list of countries and their busy and congestion tones. Users can set it to “Custom” and enter their busy and congestion tones if known.

3. Click on “**Detect**” to start PSTN detection.

Tone Settings

Busy Detection:

☒

* Busy Tone Count:

2

Congestion Detection:

☒

* Congestion Count:

2

* Tone Country:

United States of America (USA)

* Busy Tone:

f1=480@-50,f2=620@-50,c=500/500

* Congestion Tone Settings:

f1=480@-50,f2=620@-50,c=250/250

PSTN Detection:

Detect

UCM6XXX FXO Tone Settings

- If there are two analog lines connected, **Auto Detect** mode can be used.

Detect Model: Auto Detect.

Source Channel: The source channel to be detected.

Destination Channel: The channel to call during auto-detection. For example, the second FXO port.

Destination Number: The number to be dialed for detecting. This number must be the actual PSTN number for the FXO port used as the destination channel.

Dump Call Progress Tone File: Detected Call Progress Tone will be stored on a downloadable file.

PSTN Detection

Detect model:

Auto Detect

Source Channel (to be detected):

1

Destination Channel:

1

* Destination Number:

061234578

Dump Call Progress Tone

File:

Note: Detection will keep the call up for about 1 minute. If you have selected Semi-auto Detect, please follow the page prompt that pops out in the test.

Cancel

Detect

UCM6XXX PSTN Detection: Auto-Detect

If there is one analog line connected, **Semi-auto Detect** needs to be used.

PSTN Detection

Detect model:

Semi-auto Detect

Source Channel (to be detected):

1

* Destination Number:

061234578

Dump Call Progress Tone

File:

Note: Detection will keep the call up for about 1 minute. If you have selected Semi-auto Detect, please follow the page prompt that pops out in the test.

Cancel

Detect

UCM6XXX PSTN Detection: Semi-Auto Detect

Detect Model: Semi-auto Detect.

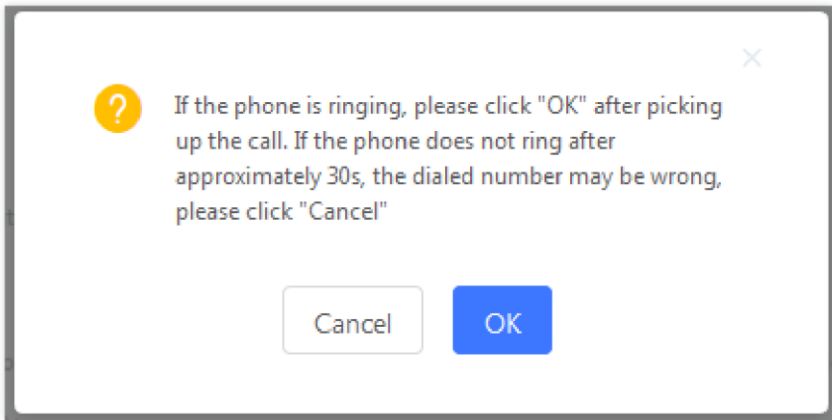
Source Channel: The source channel to be detected.

Destination Number: The number to be dialed. This number could be a cell phone number or another PSTN number that can be reached from the source channel PSTN number.

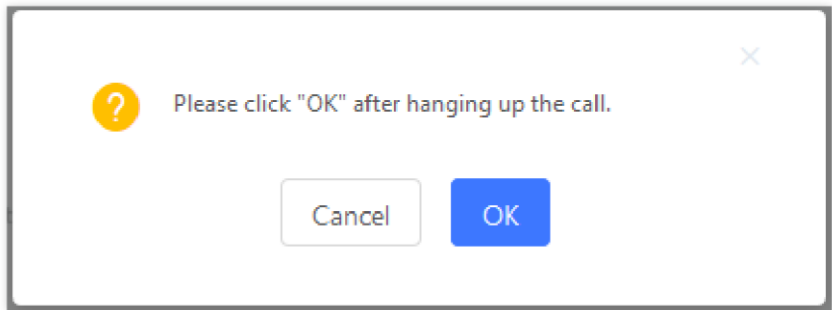
Dump Call Progress Tone File: Detected Call Progress Tone will be stored on a downloadable file.

- 1. Click "**Detect**" to start detecting.

The source channel will initiate a call to the destination number. For "**Auto Detect**", the call will be automatically answered. For "**Semi-auto Detect**", the UCM6XXX web GUI will display a prompt to notify the user to answer or hang up the call to finish the detecting process as shown in the below figures

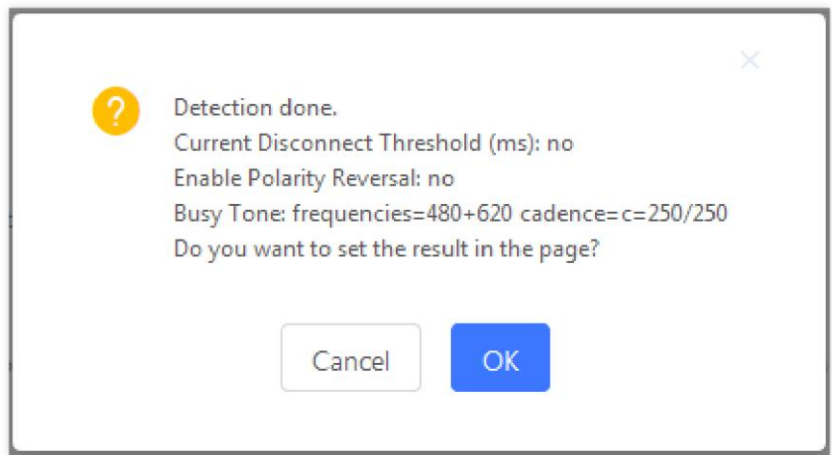


Answering Call Prompt



Hangup Call Prompt

- 2. Once done, detected results will show in "Prompt Information" window. Click on **OK** to Apply changes.



PSTN Parameters Detected

Manual Configuration of PSTN Parameters

If analog line tone parameters are known, customers could configure them on the UCM6xxx manually by setting the **Tone Country** option to **Custom** and entering them manually. The same applies for Polarity Reversal and Current Disconnect parameter.

* FXO Port

☒ 1
☐ 2

* Trunk Name

Line1

Advanced Options

Disable This Trunk

☐

SLA Mode

☐

Enable Polarity Reversal

☐

Current Disconnect Threshold (ms)

☒

200

* RX Gain

0dB

Use CallerID

☒

Caller ID Scheme

Bellcore/Telcordia

Auto Record

☐

Append #

☐

DAHDI Out Line Selection

Ascend

Echo Cancellation Mode

Default

Noise Cancelation RX Frequency (Hz)

OFF

* Ring Timeout

8000

* TX Gain

0dB

Fax Mode

None

* FXO Dial Delay (ms)

0

Direct Callback

☐

* Max Outgoing Calls

0

* Max Incoming Calls

0

Noise Cancelation TX Frequency (Hz)

OFF

Cancel

Save

Custom Tone Parameters

Otherwise, users can manually capture an analog record trace while using the Analog Trunk, to manually measure the Busy Tone frequencies and Cadence.

Please refer to the steps below for capturing an analog trace record, and manually measuring Busy Tone frequencies and cadence:

Step 1: Capture an Analog Record Trace

1. Go to **Maintenance→Signaling Troubleshooting→Analog Record Trace**.
2. Choose the FXO port used, and click on **Start** to begin the capture.
3. Make a call to FXO port, answer the call and keep it for a few seconds, then Hang-up from analog side.
4. Click on **Stop**, then **Download** to download captured audio file.

Signaling Troubleshooting

☒ Analog Record Trace
☐ Key dial-up FXO

* Record Ports:

FXO Ports:

☒ All
☒ 1
☒ 2

FXS Ports:

☐ All
☐ 1
☐ 2

Stop

Download

Delete

Output Result

Capturing...

Download Analog Trace

Note

The user can record up to 1 hour of analog record tracing.

5. Save the Downloaded file to a local folder on your computer, and unzip it.

! Users can set Record Direction to Receive since we will be analyzing received file only.

Another option is available to take analog record using A Key Dial-up FXO:

Users can directly set a PSTN number on the "External Extension" text box to troubleshoot issues related to the analog trunk easily, the following steps show how to use this feature:

1. Configure analog trunk on UCM, including outbound route.
2. Enter a reachable external number in "External Extension".
3. Press "Start" button. The call will be initiated to the external number.
4. Answer and finish the call before pressing "Stop" button.
5. The trace will be available for analysis to download after output result shows "Done! Click on Download to download the captured packets".

The screenshot shows a web interface titled "Signaling Troubleshooting". It has two radio buttons: "Analog Record Trace" (unselected) and "Key dial-up FXO" (selected). Below the radio buttons is a text input field labeled "* External Number:" containing the value "0655443322". There are three buttons: "Start" (blue), "Download" (light blue), and "Delete" (light blue). Below these buttons is a section titled "Output Result" which contains the text "The file has been deleted or does not exist."

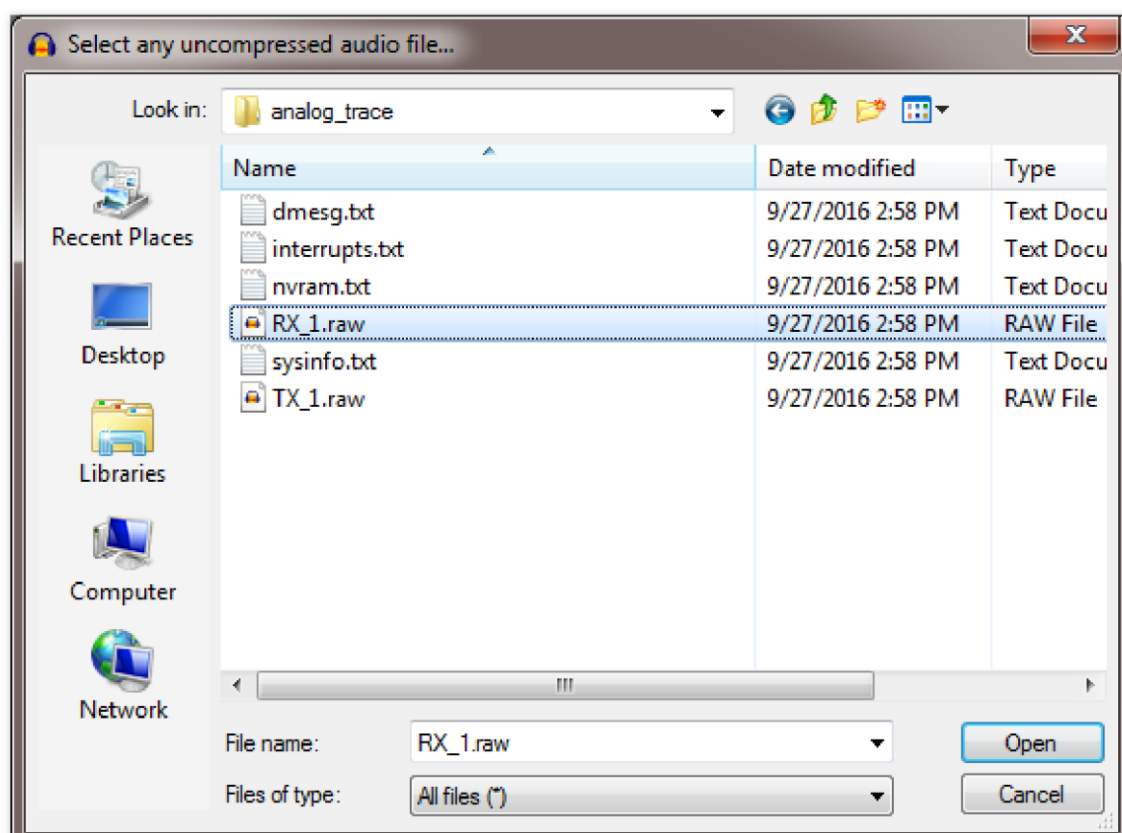
A key dial-up FXO

! When using a Key Dial-up FXO feature the outbound trunk for the analog trunk need to have internal permission. As well as it should be the trunk with the highest outbound route priority.

Step 2: Import Raw File to an Audio Analyzer Software

In this step, we will be analyzing the Analog Record Trace using Audacity. Users can download it from the following link: <http://www.audacityteam.org/download/>. Similar steps can be applied to any other Audio Analyzer Software.

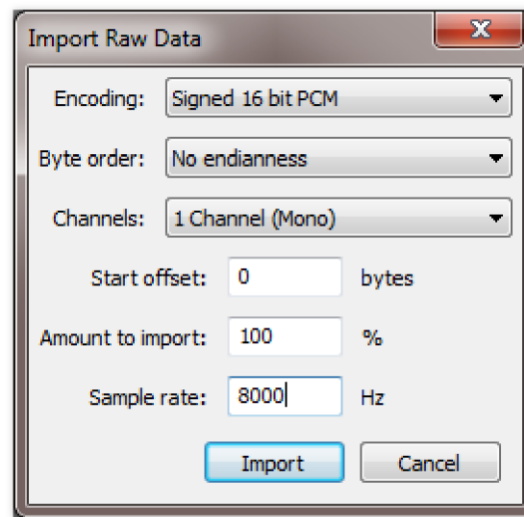
1. Import RX Raw File to Audacity.



Import Raw File to Audacity

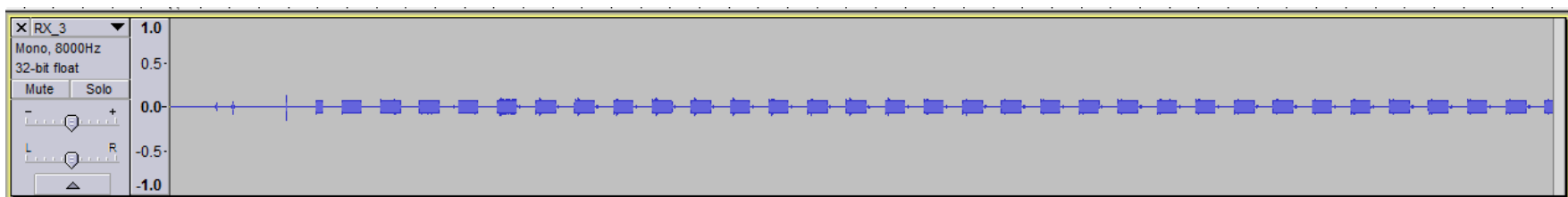
2. Click on Open, and set the following:

- **Encoding** to "Signed 16bit PCM"
- **Byte order** to "No endianness"
- **Channels** to "1 channel (Mono)"
- **Sample rate** to "8000 Hz"



Import Raw Data Parameters

3. The result will be similar to the following figure:

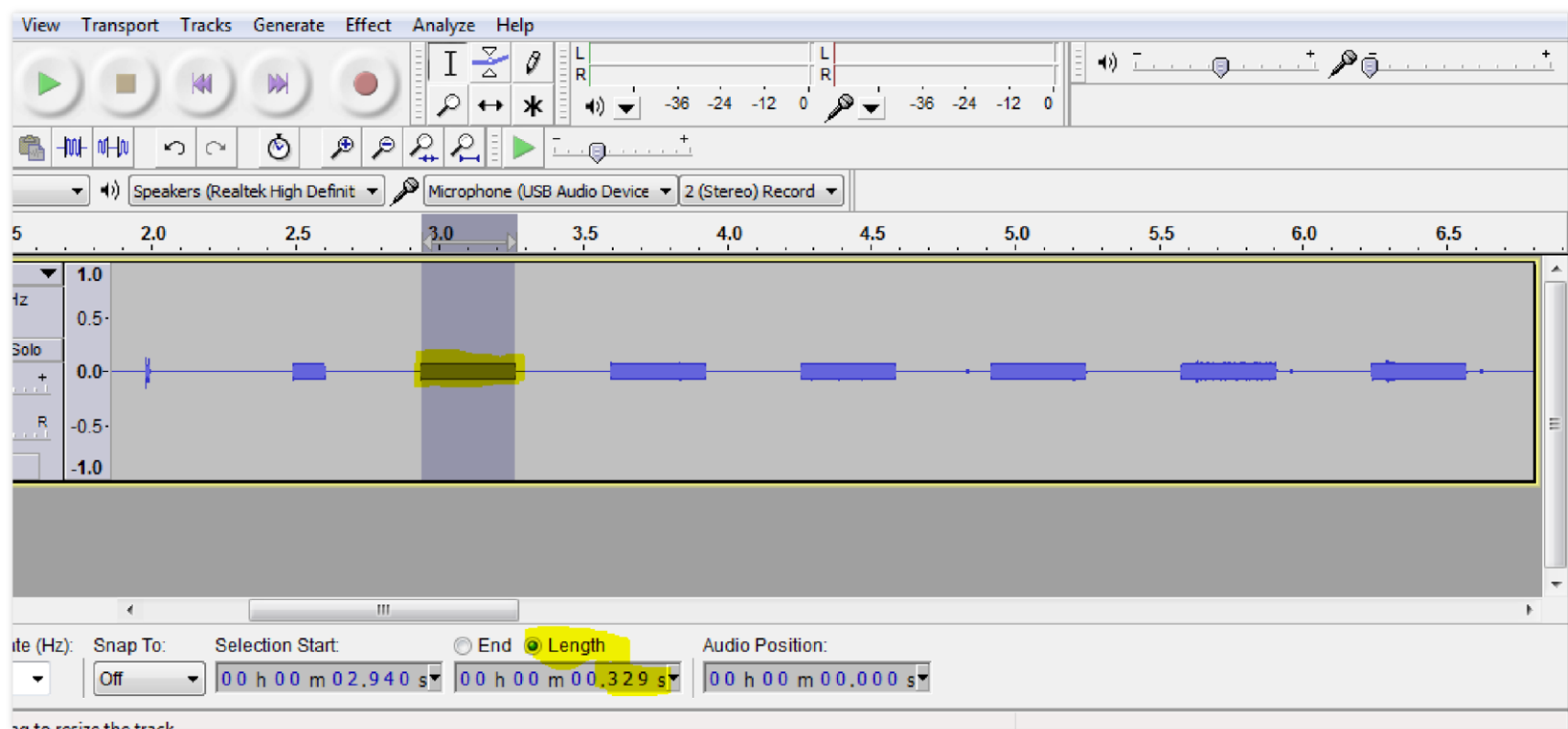


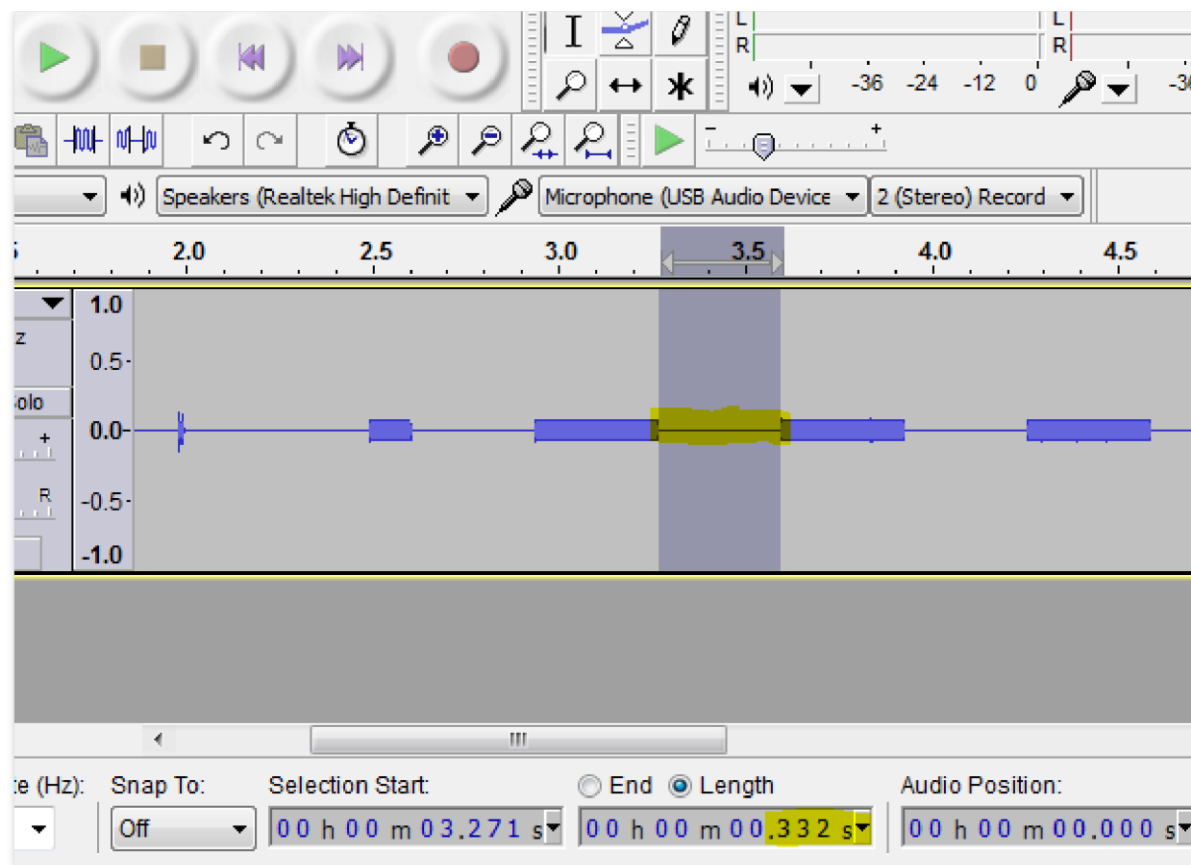
Raw Data Imported

Step 3: Measuring Cadence

Once the RX raw file is successfully imported, we can now measure ringing and silence duration that will correspond to cadence value (Cadence= Ringing duration/Silence duration).

Below is an example of how to measure silence and tone duration using the imported RX raw file.



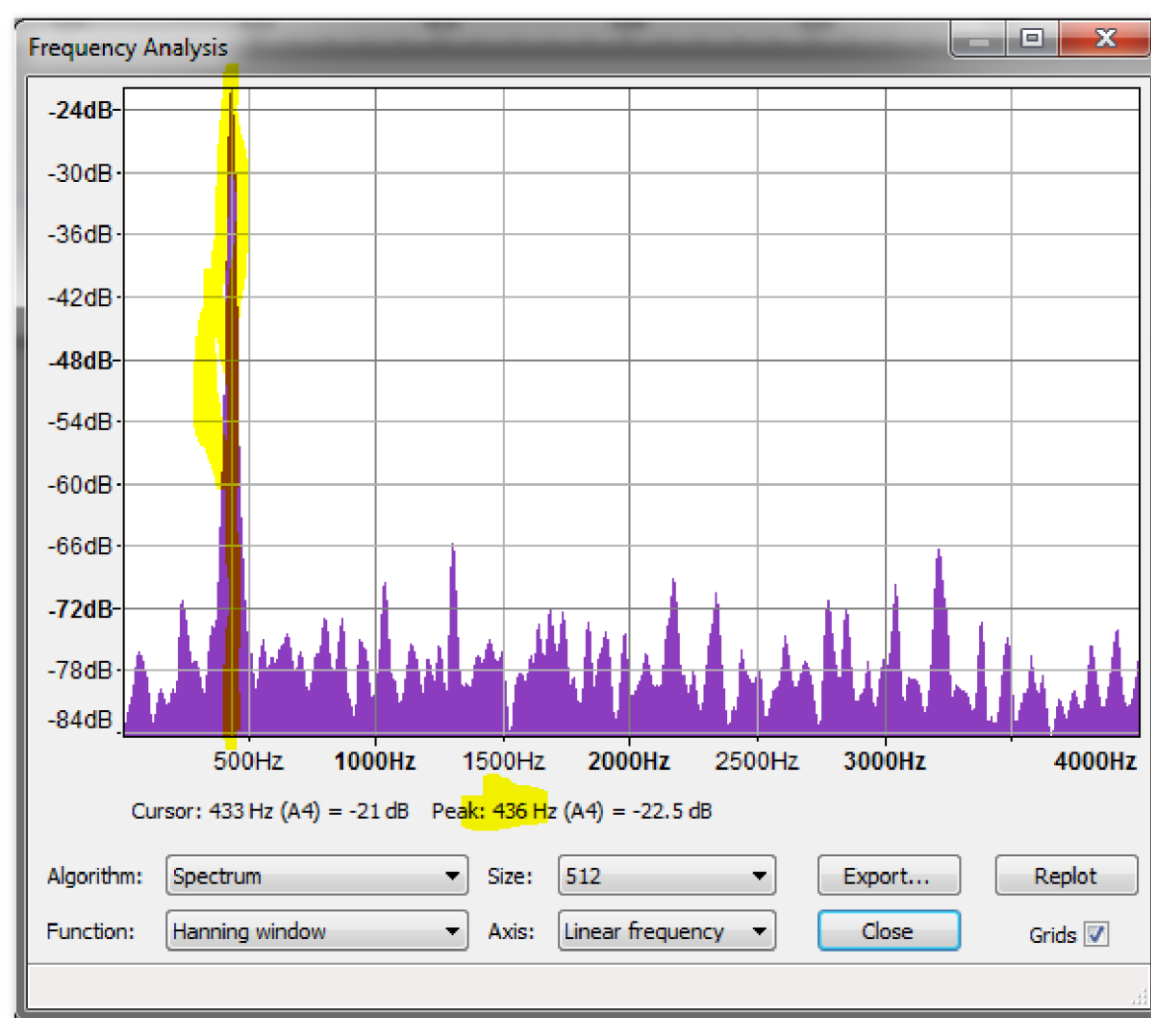


Measuring Cadence

The results are 329ms of ringing and 332 of silence, we will round them both to 330, hence the Cadence duration is $c=330/330$.

Step 4: Measuring Busy Tone Frequencies

1. On Audacity tool bar, go to **Analyze → Plot Spectrum**.
2. Point the mouse cursor on the highest signal to show the Peak Value as shown in the below figure example:



Busy Tone Frequencies

In this example we have only one frequency Peak, hence the Busy tone frequencies are $f_1=436\text{Hz}@-20\text{dB}$ and $f_2=0\text{Hz}@-20\text{dB}$. (We have rounded the attenuation value to -20dB).

Monitoring Analog Lines

The UCM6XXX supports SLA (Shared Line Appearance) that allows mapping the key with LED on a multi-line phone to different external lines. When there is an incoming call and the phone starts to ring, the LED on the key will flash in red and the call can be picked up by pressing this key. This allows users to know if the line is occupied or not. The SLA function on the

UCM6XXX is similar to BLF but SLA is used to monitor external lines i.e., analog trunks on the UCM6XXX. Users could configure the phone with BLF mode on the MPK to monitor the analog trunk status or press the line key pick-up call from the analog trunk on the UCM6XXX.

Enabling SLA Mode

1. Enable SLA Mode under the Analog Trunk Parameters.

Edit Analog Trunk: Analog1

Cancel

Save

* FXO Port:

☒ 1☐ 2

* Trunk Name:

Analog1

Advanced Options

SLA Mode:

☒

Barge Allowed:

☒

Enable Polarity Reversal:

☐

Current Disconnect Threshold

☒

200

(ms):

Hold Access:

Open

* Ring Timeout:

8000

Enable SLA Mode for the Analog Trunk

2. User can check **Barge Allowed** if needed in order to allow barging into a call in progress from SLA stations by pressing the configured line button.
3. Hold Access is used in order to allow any station to make the line on Hold, and any other station can resume it if it is set to **Open**, or restrict only that station that placed the line on hold to resume it by setting the option to **Private**.

!

SLA mode is supported on trunks with one PSTN line only. When using SLA mode, the trunk will be dedicated to SLA stations for incoming and outgoing calls bypassing configured inbound/outbound routes.

Create/Edit SLA Station

After enabling SLA Mode under the analog Trunk, SLA Station can now be configured on web GUI→**Extension / Trunk**→**SLA Station**.

SLA Station

+ Add

Delete

☐	Station Name	Station	Associated SLA Trunks	Options
No Data				

SLA Station Page

Create New SLA Station

* Station Name:

SLA_1

* Station:

2000

Associated SLA Trunks:

0 item

Available

Search

None

<

>

↑

↓

1 item

Selected

Search

Analog1

| SLA Station Options

Ring Timeout (s):

Ring Delay:

Hold Access:

Open

Create New SLA Station

- Click on “**Add**” to add an SLA Station.
- Click on  to edit the SLA Station. The following table shows the SLA Station configuration parameters.
- Click on  to delete the SLA Station.

Station Name	Configure a name to identify the SLA Station.
Station	Specify a SIP extension as a station that will be using SLA.
Associated SLA Trunks	Existing Analog Trunks with SLA Mode enabled will be listed here.
Selected SLA Trunks	Select a trunk for this SLA from the Available SLA Trunks list. Click on up and down arrows to arrange the order. If there are multiple trunks selected, when there are calls on those trunks at the same time, pressing the LINE key on the phone will pick up the call on the first trunk here.
SLA Station Options	
Ring Timeout	Configure the time (in seconds) to ring the station before the call is considered unanswered. No timeout is set by default. If set to 0, there will be no timeout.
Ring Delay	Configure the time (in seconds) for the delay before ringing the station when a call first coming in on the shared line. No delay is set by default. If set to 0, there will be no delay.
Hold Access	This option defines the competence of the hold action for one particular trunk. If set to “open”, any station could hold a call on that trunk or resume one held session; if set to “private”, only the station that places the trunk call on hold could resume the session. The default setting is “open”.

Sample Configuration

- Go to web UI→**Extension / Trunk**→**Analog Trunks** page. Create or edit the analog trunk. Make sure “**SLA Mode**” is enabled for the analog trunk. Once enabled, this analog trunk will be only available for the SLA stations created under web UI→**Extension / Trunk**→**SLA Station** page.

Edit Analog Trunk: Analog1

* FXO Port:

☒ 1☐ 2

* Trunk Name:

Analog1

Advanced Options

SLA Mode:

☒

Barge Allowed:

☒

Hold Access:

Open

Enable SLA Mode for Analog Trunk

2. Click on **“Save”**. The analog trunk will be listed with trunk mode “SLA”.

Analog Trunks

Analog Trunks

Call Progress Tone File List

+ Add

TRUNKS	DISABLE	TRUNK MODE	ANALOG PORTS	OPTIONS
Analog1	no	sla	1	

Analog Trunk with SLA Mode Enabled

3. Go to web UI→**Extension / Trunk**→**SLA Station** page, click on **“Add”**.

Users can create one or more SLA stations to monitor the analog trunk. The following figure shows two stations, 3000 and 3001 be associated with SLA trunk “Analog_Trunk”.

SLA Station

AddDelete

☐	Station Name	Station	Associated SLA Trunks	Options
☐	SLA_1	3001	Line1	
☐	SLA_2	3002	Line1	

Total: 2

<1>

10 / pageGoto

SLA Example – SLA Station

4. On a SIP Phone with extension 3000 or 3001. Configure the MPK as BLF mode and the value must be set to **“Extension_Trunkname”**, which is **3000_Analog_Trunk** or **3001_Analog_Trunk** in our example.

Multi-Purpose Keys

	Mode	Account	Description	Value
MPK 1	Busy Lamp Field (BLF)	Account 1	3000_Analog_Trunk	3000_Analog_Trunk

SLA Example – MPK Configuration

Now the SLA station is ready to use. The following functions can be achieved by this configuration:

- **Making an outbound call from the station/extension, using LINE key**

When the extension is in an idle state, pressing the line key for this extension on the phone to off-hook. Then dial the station’s extension number, in our example, dial 3000 or 3001 on the configured Phone to hear the dial tone. Then the users are able to dial external numbers for the outbound call.

- **Making an outbound call from the station/extension, using BLF key**

When the extension is in idle state, pressing the MPK and users could dial external numbers directly.

- **Answering call using LINE key**

When the station is ringing, pressing the LINE key to answer the incoming call.

- **Barging-in active call using BLF key**

When there is an active call between an SLA station and an external number using the SLA trunk, other SLA stations monitoring the same trunk could join the call by pressing the BLF key if “Barge Allowed” is enabled for the analog trunk.

- **Hold/Resume using BLF key**

If the external line is previously put on hold by an SLA station, another station that monitors the same SLA trunk could resume the call by pressing the BLF key if “Hold Access” is set to “open” on the analog trunk and the SLA station.