



Teletics Application Note

Using the Feature Server with SIP Trunks

Rev 1.0 – June 4 2012

Executive Summary

One of the key advantages of the Teletics w*intercom system architecture is the ability for the customer to integrate the w*intercom into larger networks. For example, the customer is operating a VSAT network, and wants a more seamless integration with their network. Or, where a third party VoIP provider is providing long distance services to the remote sites.

The purpose of this application note is to give a very simple example of how to integrate the Feature Server, or Feature Server RM, with a third party SIP trunking service. The intent of this document is to give a simple “where to get started” overview of how this is done. It is expected that from this simple example, someone with familiarity with SIP and IP networking could integrate a custom system with multiple accounts into a larger, working system.

SIP Trunking Overview

A SIP based VoIP provider will typically create an account with a login and password. The provider will also create a regular phone number that is serviced by this SIP account.

The Feature Server needs to incorporate four changes to use this account:

1. The Feature Server has to know about the **customers** IP network. The Feature Server that is shipped with a standard configuration does not assume that there is an external network of any kind. It is designed to only operate within the Teletics w*intercom network default design.
2. The Feature Server needs to know what to do with calls coming in via the new SIP account, and what phone on the system can access it. (Call Routing)
3. The Feature Server needs to have support for the CODEC used by the SIP provider.
4. The Feature Server needs to know how to log on to the SIP providers account (credentials, port specifications, account details).

Additional Resources

Customers attempting to integrate third party SIP provider services with the Teletics Feature Server or Feature Server RM should obtain the Teletics Feature Server Handbook.

The Teletics Feature Server Handbook is available through Teletics Support.

Should you require integration assistance from Teletics, this is available on a contracted basis, and is dependent on the amount of assistance required. Please contact your Teletics Distributor, or sales representative, for details on custom integration services.

Customer Network Settings

For the sake of our example, we will assume that the Feature Server and w*intercom 11 are connected to the LAN side of a router. The LAN side of the router must be on the network of 169.254.XXX.YYY, since the Teletics network is a zeroconfig network and uses the IP addresses of range 169.254.1.YYY to 169.254.5.YYY. The LAN side of our router has it's gateway at 169.254.5.254 in our example, but this is settable in the Feature Server. You also need to know the final DNS addresses in the network, and enter them into the Network settings page in the Feature Server:

Support:
www.teletics.com

webGUI Configuration askoziapbx.local

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Networking: Local Area Network

Local Area Network

Port	eth0 (00:0D:B9:15:C0:14)
Settings	configured manually
IP Address	169.254.5.200 / 16
Gateway	169.254.5.254
DNS Servers	75.153.176.1 75.153.176.9
MAC Address	<input type="text"/> This field can be used to modify ("spoof") the MAC address of the network interface Enter a MAC address in the following format: xxxxxxxxxx or leave blank
Topology	Public IP address
Static Public IP	<input type="text"/>

- Public IP Address: this PBX has a routable IP address (entered above)
- NAT + Static Public IP: this PBX is behind a NAT which has a static public IP. Enter this IP below.
- NAT + Dynamic Public IP: this PBX is behind a NAT which has a dynamic public IP. A hostname, constantly updated to point to this network is required. Enter this information below.

SIP Call Routing

The standard Feature Server and Feature Server RM configuration image provides 4 FXO provider ports that are configured on a “one dedicated line in/out” basis for phones 11,12,13 and 14.

In our example, we have set up the one SIP trunk provided by the third party VoIP carrier to allow the person at phone 15 to dial out using this account, and to receive calls from the number associated to this account by the third part SIP provider’s servers.

Create a new SIP provider:

The screenshot shows the Asterisk webGUI Configuration interface. The top navigation bar is red and contains the text "webGUI Configuration" and "askoziapbx.local". On the left, there is a black sidebar menu with categories: System, Connectivity, Accounts, Dialplan (print), Services, and Status. The "Accounts" category is expanded, and "Providers" is highlighted with a red circle. The main content area is titled "Accounts: Providers" and features a tabbed interface with "SIP", "IAX", and "Analog" tabs. The "SIP" tab is selected and circled in red. Below the tabs, there is a table titled "Analog" with the following data:

Name	Pattern(s)	Number	Port
Analog FXO 1 J4	9 .	11	Port 4
Analog FXO 2 J3	9 .	12	Port 3
Analog FXO 3 J2	9 .	13	Port 2
Analog Port 4 J1	9 .	14	Port 1

A new SIP provider menu window will open. Here is an example of typical SIP credentials that you would need to enter. First, the Main Account information and Routing:

The screenshot displays a configuration interface for a SIP provider. On the left is a dark sidebar menu with categories: Storage, Connectivity, Accounts, Dialplan (print), Services, and Status. The main area is divided into two sections: 'General Settings' and 'Call Routing'. In 'General Settings', fields for Name, Host, Username, Password, and Public Number are highlighted with red circles. In 'Call Routing', the 'Outgoing Patterns' field is highlighted with a red circle.

General Settings	
Name	Powersat_Test <small>Descriptive name for this provider.</small>
Host	ep.asteriskrgns.net : 15061 <small>Provider host URL or IP address and optional port.</small>
Username	200110125422
Password	abc123
Language	English (US) <small>Audio prompts will be played back in the selected language for this account.</small>
Public Number	7808336500 <small>This 'external' number will be read back to the caller when reaching voicemail; defaults to account's username if it is numeric. If it is not, the internal extension this call was routed to will be read back.</small>

Call Routing	
Outgoing Patterns	9. <small>Enter patterns, one per line, to define this provider's outgoing routing. Enter "X!" to use this provider for all outgoing calls. To access this provider by using a "9" prefix, enter "9!". If no patterns are entered, only incoming calls will be accepted.</small>

- + - adds a prefix (i.e. "1+555" matches "555" and passes "1555" to the provider)
- | - removes a prefix (i.e. "1|555" matches "1555" but only passes "555" to the provider)
- X - matches digits 0-9
- Z - matches digits 1-9
- N - matches digits 2-9
- [13-5] - matches any digit in the brackets (here, 1,3,4,5)
- . - matches one or more characters (not allowed before | or +)
- ! - matches zero or more characters (not allowed before | or +)

In this example:

- The **Name** setting is whatever you want to call this SIP provider/account.
- The **Host** URL (and port if needed), **Username**, **Password**, and **Public Number** is provided by your SIP provider/carrier for the account you are using.
- The Call Routing example specifies that when processing the call, we use 9 as the lead digit to indicate a SIP call, and then drop the lead 9 before passing the dial string to the SIP provider for further routing.

Setting Codecs

Your SIP provider will specify what CODEC(s) they use. You have to ensure that these are enabled in the Feature Server:

	Enabled (drag-and-drop)	Disabled
Audio	G.729A	G.722
	G.711 u-law	G.723.1
	G.711 A-law	G.726 (AAL2)
	GSM	G.726 (RFC3551)
		iLBC
		SpeeX
		ADPCM
		16 bit Signed Linear PCM
		LPC10
Video	Enabled (drag-and-drop)	Disabled
		H.261
		H.263
		H.263+
		H.264

Please note that the Codec G.711 u-law has to be left in the enabled menu to ensure that the w*intercom system works properly.

Advanced SIP Account Settings

Some SIP providers require additional security information from the user when registering. An account name/identifier is a typical requirement. You can have the Teletics Feature Server send this information automatically on logon by entering the account number in the **From User** field in the **Advanced** settings:

Advanced Options

Registration Do not register with this provider.
Completely override the generated registration string with a manually defined one.

Qualify 2 seconds
The provider will be contacted this often to check its availability.

NAT always use NAT mode.
Is there a NATing device between the PBX and this account?

DTMF Mode auto

Authorization User
Some providers require a separate username for authorization. Defaults to username entered above.

From User 200110125422
Some providers require a separate 'from' user. Defaults to username entered above.

From Domain
Some providers require a separate 'from' domain. Defaults to host entered above.

Manual Attributes
Manual key-value pairs can be entered in addition to the generated configuration (i.e. configuration=value). These settings will be appended to this item's Asterisk configuration file or

Once you have these settings entered, you will notice a SIP provider has been added on the providers main screen. You can go back and edit these settings at any time later:

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www.teletics.com

webGUI Configuration askoziapbx.local

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Accounts: Providers

SIP IAX Analog

SIP

Name	Pattern(s)	Username	Host
Powersat_Test	9 .	200110125422	ep.asterisk.rgns.net

Analog

Name	Pattern(s)	Number	Port
Analog FXO 1 J4	9 .	11	Port 4
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Enabling SIP Provider Account

All phones on the system can be told to use (or not use) certain providers. You need to check that the w*intercom phone you want to have use the account you just created is allowed to do so in the w*intercom phone settings.

From the main menu, select Phones, and then click on the little gear symbol next to the phone you want to use the account you just created. In our example, this will be w*intercom phone 15:

Check that the provider you just created is not blocked:

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webGUI Configuration

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Accounts: Phones

SIP IAX External

Extension	Caller ID	Description	
11	T11	Phone 11	
12	T12	Phone 12	
13	T13	Phone 13	
14	T14	Phone 14	
15	T15	Phone 15	
16	T16	Phone 16	
17	T17	Phone 17	
18	T18	Phone 18	
19	T19	Phone 19	
20	T20	Phone 20	
21	T21	Phone 21	
22	T22	Phone 22	
23	T23	Phone 23	
24	T24	Phone 24	

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ACCOUNTS: EDIT SIP PHONE

General Settings

Number
The number used to dial this phone. Use this number as your username.

Caller ID
Text to be displayed for Caller ID.

Language
Audio prompts will be played back in the selected language for this account.

Ring Length
The number of seconds this phone will ring before giving up or going to voicemail.

Description
You may enter a description here for your reference (not parsed).

Security

Password
This account's password.

Public Access allow this number to be reachable over the Internet

Block Providers

- Analog FXO 1 J4
- Analog FXO 2 J3
- Analog FXO 3 J2
- Analog Port 4 J1
- Powersat_Test

Block access to the providers selected above.

Basic Testing and Troubleshooting

First, if the Feature Server is properly getting a time setting, the **Network** settings are correct. If the date stays on January 1, you **MUST** fix your **Network** settings prior to anything else. Here is an example of the first few lines of the log file following a reboot. Notice that the time is correctly set in the first few lines:

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webGUI Configuration askoziapbx.local

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Status: Logs

System log entries

Jan 1 00:00:15 syslogd started: BusyBox v1.15.3
Jan 1 00:00:18 ntpclient[1265]: Using server pool.ntp.org
Jun 3 13:38:29 ntpclient[1265]: Time set from remote server via settimeofday()
Jun 3 13:38:29 ntpclient[1265]: skew 59.2045090917.2ms, freq adjust 0
Jun 3 13:38:29 syslog: Password for 'root' changed
Jun 3 13:38:29 dropbear[1284]: Running in background
Jun 3 13:38:29 asterisk[1308]: VERBOSE[1301]: Asterisk Dynamic Loader Starting:
Jun 3 13:38:29 asterisk[1308]: VERBOSE[1301]: == Parsing '/etc/asterisk/modules.conf':
Jun 3 13:38:29 asterisk[1308]: VERBOSE[1301]: == Found
Jun 3 13:38:29 asterisk[1308]: VERBOSE[1301]: == Manager registered action Ping
Jun 3 13:38:29 asterisk[1308]: VERBOSE[1301]: == Manager registered action Events
Jun 3 13:38:29 asterisk[1308]: VERBOSE[1301]: == Manager registered action Logoff
Jun 3 13:38:29 asterisk[1308]: VERBOSE[1301]: == Manager registered action Login
Jun 3 13:38:29 asterisk[1308]: VERBOSE[1301]: == Manager registered action Challenge
Jun 3 13:38:29 asterisk[1308]: VERBOSE[1301]: == Manager registered action Hangup
Jun 3 13:38:29 asterisk[1308]: VERBOSE[1301]: == Manager registered action Status
Jun 3 13:38:29 asterisk[1308]: VERBOSE[1301]: == Manager registered action Setvar
Jun 3 13:38:29 provisioning_configure(): pid finder returned:
Jun 3 13:38:29 provisioning_configure(): No existing httpd provisioning server running, not killing.

When connecting to your SIP provider, you will also see a log entry that will show if an attempted connection fails. The Feature Server will also retry on an ongoing basis. Failed attempts will show the number of failed tries as well.

If SIP logins continue to fail and all your settings appear to be correct, it is recommended that you contact your SIP provider and ask for assistance.