

# Ecessa Proxy VoIP Manual

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## Introduction

The Ecessa VoIP proxy is designed to act as a proxy for SIP calls. The VoIP proxy will rewrite headers so that the provider sees the public WAN addresses instead of the LAN IPs of the phones/PBX. The VoIP proxy also by default will attempt to failover calls it determines have issues whether that is due to traffic failure or other criteria.

## Configuration Overview

The Ecessa VoIP Proxy can be configured through the web user interface via **Advanced Setup : Configure VoIP**

The Ecessa VoIP Proxy by default will use all the available WANs on the device. This will load balance registrations and outbound calls on the WANs specified in the **VoIP WAN Configuration**. If there is a WAN that should be used as the primary WAN for VoIP then the associated primary checkbox should be checked.

Besides the WAN configuration on the VoIP Proxy most of the other default settings should remain the same. Please note that the SIP/RTP ports used by the Ecessa do not have to match with what the phones and/or provider is using, thus the default port settings should be left alone. The only time the RTP range should be modified is if the number of concurrent calls at any one time will be greater than 500 calls. The default RTP range is 7000 - 7999. A single concurrent call takes 2 ports, one for RTP and one for RTCP which contains statistics about the RTP.

[Configuration Reference](#)

## VoIP failover requirements

When the VoIP proxy determines that a call must be failed over to a different WAN line it will perform a Re-Invite where it will change the IP associated. If the provider does not support this or has that functionality disabled then the failover will not work.

[Re-Invite with IP changes is allowed but requires re-authentication](#)

When this is the case the VoIP proxy must contain the authentication information for the phones. To do this go into the top link on the configuration page called **Configure VoIP Phone Accounts**

Add a domain where the domain name is what will be used as the **Authentication Realm**. This can be retrieved from the provider.

In the example below the **Domain Name** is ProviderRealm.

```
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 1.2.3.4
From: "User A" <sip:6000@myprovider.com>
To: <sip:userB@myprovider.com>
```

WWW-Authenticate: DIGEST realm="ProviderRealm",  
nonce="hie7ieniejaiWeeS3doh0air5rieth", algorithm=MD5

After activating the settings click configure to add users to the new domain. If there is a global password for all the users then all that is necessary is to fill that in to the **Global Password field**.

## Import VoIP Authentication

To import select a file to import and click **Import VoIP Authentication**.

### Requirements

- The file to import must be a CSV file (.csv)
- There must be exactly 3 columns in the CSV provided
  - Line Name - Should match the **From** user in the SIP message.
  - Authentication Username
  - Authentication Password
- The first line in the CSV must be a header line. This line will be ignored
- Below is an example of a CSV file that can be imported.  
Line Name,User Name,User Password  
6000,6000-username,6000-password  
6001,6001-username,6001-password

## Add a user manually

Add an entry similar to what is specified below. Below are the specifics of the user we wish to add.

- The Line Name is **6000**
- The Authentication User Name is **6000-user**

Entry	Line Name	Disabled	Description	Username	Password	Delete
1	6000	<input type="checkbox"/>		6000-user	*****	Delete

In the example below the Line Name is 6000.

SIP/2.0 401 Unauthorized

Via: SIP/2.0/UDP 1.2.3.4

From: "User A" <sip:6000@myprovider.com>

To: <sip:userB@myprovider.com>

WWW-Authenticate: DIGEST realm="ProviderRealm",

nonce="hie7ieniejaiWeeS3doh0air5rieth", algorithm=MD5

## Setup

### Hosted setup

The most basic use of the VoIP proxy is where there are multiple phones on the LAN that are communicating directly with the provider. In this setup the phones and the provider need to know nothing about the VoIP proxy since it will be acting in transparent mode.

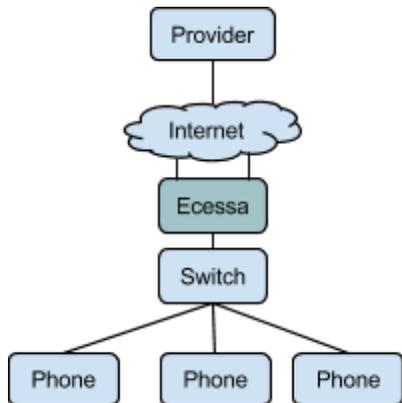
## Example network configuration

DHCP LAN with **192.168.1.1/24** network which have that SIP phones on it.

The provider destination port is **5060**.

There are **2 WANs** configured for the VoIP proxy.

## Diagram



## Configuration

Enable **SIP Redirect** and add an entry that looks like the following:

Entry	Source Address	Source Port	Destination Port	LAN Alias	Disable	
1	192.168.6.1/24	Any	5060	LAN	<input type="checkbox"/>	Delete

When the phones boot they will go out to the provider and retrieve their configurations. The phones will not know that there is a proxy in between. The VoIP proxy in this case will rewrite the headers of the phones and send them to the provider for anything that it sees that comes from the LAN destined to port 5060.

## PBX on the LAN

If the phones on the LAN communicate to a PBX and the PBX in turn communicates to the provider then the VoIP proxy should be configured as follows.

## Example network configuration

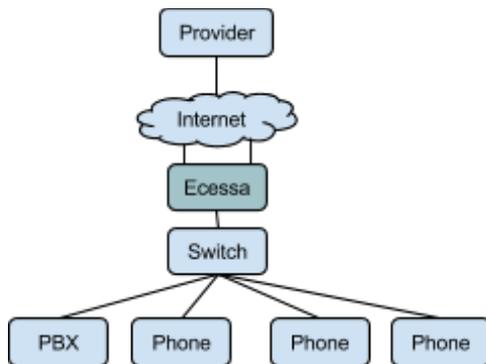
DHCP LAN with **192.168.1.1/24** network which have that SIP phones on it.

The PBX IP address is **192.168.1.100**.

The provider destination port is **5060**.

There are **2 WANs** configured for the VoIP proxy.

## Diagram



## Configuration

Enable **SIP Redirect** and add an entry that looks like the following:

Entry	Source Address	Source Port	Destination Port	LAN Alias	Disable	
1	192.168.6.1/24	Any	5060	LAN	<input type="checkbox"/>	Delete

Also add an inbound proxy for the PBX. This is located on the **Advanced** tab.

<input checked="" type="checkbox"/> Enable Inbound Proxy								
Entry	Proxy Protocol	User Name	Proxy Host	Proxy Port	LAN Alias	NAT Disable		
1	UDP		192.168.1.100	5060	LAN	<input type="checkbox"/>	<input type="checkbox"/>	Delete

With the configuration above the VoIP Proxy will proxy all LAN traffic that is going outbound to the destination port of 5060. Also when inbound calls come in that does not match the registration that we have seen we will send it to the PBX.

## Quality

The Ecessa **Bandwidth Throttling (QoS)** feature should be setup to ensure the audio quality of phone calls is as good as it can be. Outbound and Inbound QoS should be configured for each VoIP WAN configured based on the number of concurrent calls expected. The bandwidth required for the audio depends on the codec being used.

For example a common codec used with VoIP providers is G.711 which uses 87.2 Kbps for bandwidth. So if you expect to have 100 concurrent calls your bandwidth should be:  
 $87.2 \text{ Kbps} * 100 = 8.720 \text{ Mbps}$ .

If you are unsure how much VoIP traffic will be generated it is recommended to start with 50% of the bandwidth on a WAN line reserved for VoIP. If bandwidth is available due to lack of VoIP traffic, other services can utilize up to 100% of the bandwidth.

## Configure

Create a classifier rule called **voip**. Add 1 entry that looks like the example below and click **Activate**.

#	SrcIP/Mask	DstIP/Mask	Protocol	Srcport Begin:End	Dstport Begin:End	Length Min:Max	Session Min:Max	DSCP	
1			RTP	0:0	0:0	0:0	0:0		Delete

For each WAN configure both the Inbound and Outbound QoS to use this classifier.

Enable Rate Limited Classes on WAN1

#	Queue	Parent	Class	Priority	Rate (% k)	Max (% k)	Classifier	
1	fifo	root	voip	1	50%	100%	voip	Delete

Once the QoS for VoIP has been configured, the VoIP page should show the WANs with a QoS value.

Entry	Status	Alias	IP	Public IP	QOS-OUT	QOS-IN	Primary	Disable	
1	UP	WAN1	100.1.1.1		50 %	50%	<input type="radio"/>	<input type="checkbox"/>	Delete

## Call Statistics

Calls that are performed on the Ecessa device can be seen via the **Call List** tab on the VoIP page. This will allow you to see the current calls as well as search for historical calls.

## Search Options

There are a lot of criteria that is available to search on.

The following will search for all in progress calls are destined to a user that contains 6000 in its username.

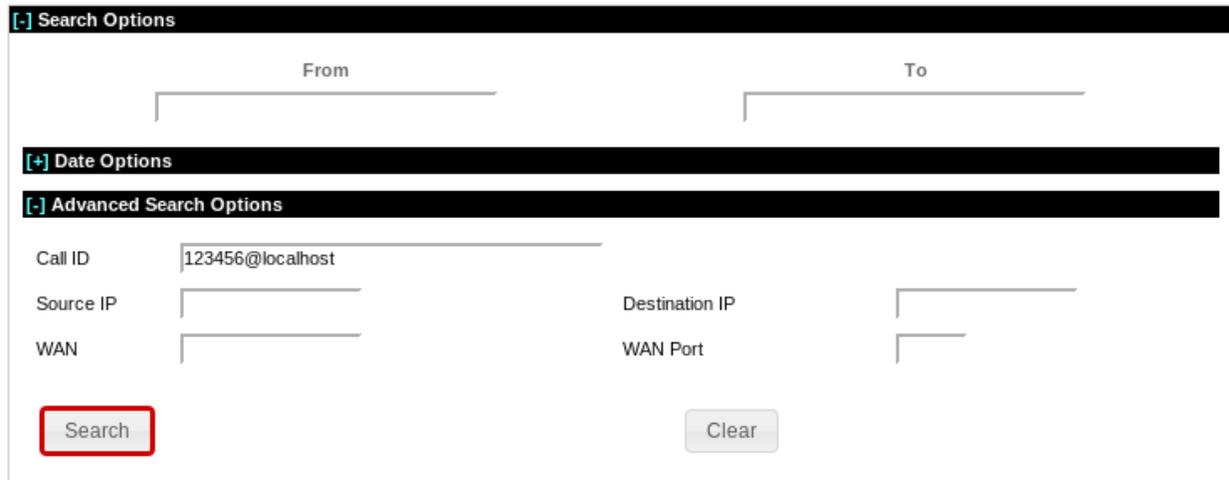
**[ - ] Search Options**

From \_\_\_\_\_ To 6000

**[ + ] Date Options**

**[ + ] Advanced Search Options**

Search by Call-ID is very useful. The following example searches for all in progress calls that match the Call-ID **123456@localhost**



The screenshot shows a search interface with the following elements:

- [-] Search Options**: A header bar with a minus sign icon.
- From** and **To**: Two empty text input fields.
- [+] Date Options**: A header bar with a plus sign icon.
- [-] Advanced Search Options**: A header bar with a minus sign icon.
- Call ID**: A text input field containing "123456@localhost".
- Source IP**: An empty text input field.
- Destination IP**: An empty text input field.
- WAN**: An empty text input field.
- WAN Port**: An empty text input field.
- Search**: A button with a red border, highlighted.
- Clear**: A button with a grey background.

## Call List

When either the search criteria is filled in or the **Show Latest Calls** is pressed a list of calls based on the search criteria will show up. This will contain basic information about the call. To view more information about a call click **View** button.

## View a call

When a call is viewed, a new separate window will show the information associated to the call.

### Basic Information

This section contains connection information about the call as well as the start date and duration of the call. If the call is in progress the duration will show as **In Progress**.

### Advanced Information

By default this section is collapsed. This section is useful when troubleshooting a call.

### Statistics

This section contains the quality of the call. The **Current** column will show different information depending whether the call is in progress or the call has ended.

- If the call is in progress then the values are based on the last time the RTCP was retrieved for the specific direction.
- If the call has ended the values are based on the average quality during the call.

If the quality section contains N/A for 1 or both directions then the RTCP information has not yet been retrieved. This is usually due to the provider and/or phone not sending RTCP information.

If one or both sides do not have quality information then the latency information will also not be shown because the latency information needs RTCP information from both directions.

## Graphs

When a call is in progress a live graph will be shown of the current quality information.

## Call Ladder

This section shows what happened to a call. This will always show what WAN the call connected on as well as the SIP messages that were sent through the Ecessa VoIP proxy. Below is an example of a simple call.

[-] Call Ladder <input type="button" value="Refresh"/>				
Date	asterisk	ClariLink	900	
08/18/2014 15:06:54.617887	----- WAN: 100.1.2.2:5060			
08/18/2014 15:06:54.617887	192.168.6.106:5060	--INVITE-->		100.1.1.223:5060
08/18/2014 15:06:54.941377	192.168.6.106:5060		<--Trying--	100.1.1.223:5060
08/18/2014 15:06:54.941939	192.168.6.106:5060		<--OK--	100.1.1.223:5060
08/18/2014 15:06:54.988860	192.168.6.106:1024	--ACK-->		100.1.1.223:5060

Below is an example of a call failover event. This shows the original call initiated on WAN 100.1.2.2 and then about 32 minutes later a failover occurred onto WAN 200.1.2.2. During the failover all communication is from the ClariLink to the provider.

[-] Call Ladder <input type="button" value="Refresh"/>				
Date	asterisk	ClariLink	900	
08/18/2014 15:06:54.617887	----- WAN: 100.1.2.2:5060			
08/18/2014 15:06:54.617887	192.168.6.106:5060	--INVITE-->		100.1.1.223:5060
08/18/2014 15:06:54.941377	192.168.6.106:5060		<--Trying--	100.1.1.223:5060
08/18/2014 15:06:54.941939	192.168.6.106:5060		<--OK--	100.1.1.223:5060
08/18/2014 15:06:54.988860	192.168.6.106:1024	--ACK-->		100.1.1.223:5060
08/18/2014 15:38:42.788464	----- WAN: 200.1.2.2:5060			
08/18/2014 15:38:42.788464	CL60		--INVITE-->	100.1.1.223:5060
08/18/2014 15:38:42.790202	CL60		<--Request Pending--	100.1.1.223:5060
08/18/2014 15:38:42.793400	CL60		--ACK-->	100.1.1.223:5060
08/18/2014 15:38:42.796916	CL60		--INVITE-->	100.1.1.223:5060
08/18/2014 15:38:42.832168	CL60		<--Trying--	100.1.1.223:5060
08/18/2014 15:38:42.870126	CL60		<--OK--	100.1.1.223:5060
08/18/2014 15:38:42.870330	CL60		--ACK-->	100.1.1.223:5060

## Trace Route

At the top of the **VoIP Call Information** page there is a **Trace Route** tab. When a call is established a traceroute is performed over the WAN the call established on. This shows When a call is initiated a trace route is performed over the WAN the call is on.

# Security

It is recommended to put the provider possible SIP addresses into the **Allowed SIP Sources** on the **Advanced** tab in order to prevent Denial Of Service (DOS) attacks on the WAN SIP ports. Having this section filled in ensures that only the provider can send SIP to the Ecessa VoIP proxy. Each entry in this section can take either an IP, IP/NM, or FQDN.

Allowed SIP Sources (3/16)

Enable Allowed Sources

Entry	Source Host or IP	Disable	
1	<input type="text" value="100.1.1.100"/>	<input type="checkbox"/>	<input type="button" value="Delete"/>
2	<input type="text" value="100.1.1.100/24"/>	<input type="checkbox"/>	<input type="button" value="Delete"/>
3	<input type="text" value="voip.myprovider.com"/>	<input type="checkbox"/>	<input type="button" value="Delete"/>

# Troubleshooting

## Not proxying

The following problems are the most common and are good indications that the Ecessa VoIP proxy has not been configured properly and it not proxying the SIP.

- Phones will not register
- 1 way audio or no audio
- Calls are not showing up in the **Call List**

Try the following to fix the issues above:

- Make sure that the **Static Routes** and **One-to-One NAT** does not have conflicting address information that matches what the **VoIP** proxy is using. If there are conflicting addresses then disable the associated entries on those features.
- Verify that the **SIP Redirect** entry is correct.
- Contact Ecessa Technical Support

# References

[VoIP KB articles](#)