

# Advanced SIP recording for small to mid-size organisations

## **Product Documentation**

Last Updated: February 2016

## **Table of Contents**

1. Home	3
1.1 Legal & copyright notices	3
1.2 What is Echo?	4
1.3 System requirements	4
1.3.1 ISDN/Analogue	4
1.3.2 VOIP	4
1.4 ISDN call recording 5	5
1.4.1 USB devices	5
1.4.2 PCI cards $\ldots$ 6	3
1.4.3 Array	3
1.5 SIP/VoIP call recording	Э
1.5.1 Port mirroring	Э
1.5.2 SCCP/SIP extension-side configuration 1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.	10
1.5.3 SIP trunk configuration 1	10
1.6 Echo setup 1	11
1.6.1 Installing Echo 1	11
1.6.2 Configuring Echo 1	13
1.7 Echo interface 1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.	15
1.8 Echo integration 1	18
1.8.1 Overview	18
1.8.2 Integrating with TIM Plus	19
1.8.3 Integrating with TIM Enterprise	21
1.8.4 Acessing your call recordings	23
1.9 Call obtuscation (PCI-DSS)	24
1.9.1 Programmatic obfuscation	25
1.9.2 DIME obtuscation	27
1.9.3 Call obtuscator 2	28

## Home

### Legal & copyright notices

#### **Software license**

When you purchase this software, you are actually purchasing a license to use it.

### **Disclaimer**

Tri-Line Network Telephony Ltd (hereafter named "Tri-Line") makes no warranties nor representations (neither expressed nor implied) with respect to the contents or performance of the product or this documentation. It particularly disclaims any warranty of fitness or merchantability for any particular purpose.

The product is sold "as is" with any faults. Any claims made by sales literature or salespersons do not constitute warranties.

Because of the diversity of hardware, software and conditions under which the system may be used, Tri-Line cannot make any warranty of fitness for a particular purpose. The entire risk of using the product must be assumed by the user. Accordingly, the user is recommended to thoroughly test the product before relying on it. In any event, any liability of Tri-Line is limited exclusively to a refund of the purchase price of the product.

It is the user's responsibility to ensure that the product or its use conforms to any laws concerning the provision of data protection in their organisation.

Tri-Line reserves the right to revise and make changes to the software and/or the hardware and/or this documentation without incurring any obligation to notify any person of such changes and/or revisions.

By using the software you agree to be bound by these terms and conditions.

### Copyright

A

Echo ® is a registered trademark of and copyright © Tri-Line Network Telephony Limited, London, England, 2013.

All rights of the manufacturer are reserved. Any unauthorised lending, copying, hiring, or any other form of distribution, electronically or otherwise, without the consent of the copyright holders is strictly prohibited.

The contact details of the copyright holders are:

Tri-Line Network Telephony Limited 9-10 Telfords Yard The Highway London E1W 2BS

Switchboard: +44 20 7265 2600 Technical Support: +44 20 7265 2626 Website: http://www.tri-line.com/

### **Free upgrades**

We operate a free upgrade scheme for customers who purchase maintenance at the same time as purchasing a license; whilst a maintenance contract is in place, minor software updates and enhancements are made available free of charge.

Free upgrades are solely at the discretion of Tri-Line and are usually delivered by electronic means over the internet. It is the customer's responsibility to ensure that these updates can be received.

Customers without a maintenance contract will be charged for any software upgrades they require, as well as for any technical assistance needed during the upgrade procedure.

### What is Echo?

Echo is the call recording module that can be incorporated into one of our call logging solutions, TIM Plus or Enterprise in order to provide a fully integrated call logging and recording solution. It can be attached to either ISDN or SIP lines and, any phone calls going over those channels are intercepted, recorded, then sent to our call logger to be attached to the logged call.

To record calls over PSTN, channels such as ISDN30 (PRI/E1/T1/J1), ISDN2 (BRI) or analogue POT lines, a piece of physical hardware is used to connect into your lines, such as a USB box or a PCI card, depending on your deployment needs.

For VoIP (SIP) channels, a PC with a standard network interface card (NIC) can be used to capture the voice packets from strategic points in your voice network. If many simultaneous VoIP calls are expected, a dedicated computer with multiple NICs becomes necessary.

All calls are securely encrypted using the industry standard 256-bit key AES algorithm.

### System requirements

### **ISDN/Analogue**

The PC hardware requirements are determined by the size and complexity of the devices that you want to attach to the system.

#### **Small deployments**

For smaller deployments of under 30-channels PRI/SIP, 8-Channels BRI or 8-channels of POTS/Analogue, a relatively low-specification PC can be used:

- Windows 7/8/8.1/10, Windows Server 2008 R2 or Windows Server 2012
- Dedicated USB hub and controller (Only with Echo USB, Echo cards will have a RJ45 connection on the back so not required.)
- 4GB RAM
- Intel Core i5-2320 3Ghz or equivalent
- PC should not be used for other heavy-CPU tasks

#### **Medium-sized deployments**

For single-site deployments up to 240 ISDN channels, the following PC specification is recommended:

- Windows 7/8/8.1/10, Windows Server 2008 R2 or Windows Server 2012
- PCI/PCI-Express slots for Echo hardware
- 8GB RAM
- Intel Dual-Core i5 3Ghz or equivalent
- · Keyboard, and mouse (not necessary for Windows 8)
- PC must be a dedicated machine

#### Large deployments/arrays

For larger deployments, our Helpdesk team will confirm hardware specifications with you.

### VOIP

In order for Echo to collect call audio information, it must be installed on a computer that has an additional network card connected to a mirrored port on a local managed network switch.

The primary network card would have its IP configuration set up so that it is visible on the data LAN. This allows playback of recordings by authorised users, as well as remote control of the Echo server, such as Remote Desktop Connection, if required.

The secondary network card would have its IP configuration set up so that it is visible on the SIP/telephone network. This allows mirrored traffic to be sent to it for the purpose of monitoring and recording by the Echo server.

Field	Description	
CPU	minimum 4 cores, 2GHz	
Memory	3GB	
Hard disk	250GB Hard disk space = 44,000 channel hours	
	500GB Hard disk space = 88,000 channel hours	
	700GB Hard disk space = 132,000 channel hours	
	1TB Hard disk space = 176,000 channel hours	
Operating system	Windows XP SP3 - Windows 8 (inc. Server editions)	
Physical adaptors Realtek PCI GBE Family Controller		
	D-Link DGE-528T Gigabit Ethernet Adapter,	
	Realtek PCIe FE Family Controller, Intel(R) 82566DM-2 Gigabit	
	Network Connection, Intel(R) 82576 Gigabit Dual Port Network,	
	TRENDnet, TEG-PCITXR 32-bit 10/100/1000Mbps PCI ADAPTER	
	Intel(R) PRO/1000 PM Network Connection,	
	AMD PCNET Family PCI Ethernet Adapter,	
	HP NC112i 1-port Ethernet Server Adapter,	
	Intel(R) Gigabit CT Desktop Adapter,	
	Broadcom NetXtreme Gigabit Ethernet	
Audio codecs	G.711 a-law	
	G.711 µ-law	

### **ISDN call recording**

In addition to the Echo software, to record calls over PSTN, channels such as ISDN30 (PRI/E1/T1/J1), ISDN2 (BRI) or analogue POT lines, a piece of physical hardware is necessary to physically tap into your lines. These hardware devices passively intercept your phone lines, monitoring and recording all calls that pass along the lines. The hardware is available in various form factors, in order to address various deployment needs.

For small deployments of 30-channels PRI/SIP, 8-Channels BRI or 8-channels of POTS/Analogue, a USB box is recommended. For medium-sized deployments of up to 240 ISDN channels, a PCI or PCI-Express card will be required, and for sites with a larger channel count, we tend to supply a HyperArray Voice Recording Unit.

### **USB devices**

The Echo USB Voice Recording devices is suitable for small to medium channel count and can be attached to any telephone system connected to one of our call logging solution, TIM Plus or Enterprise. The USB box will be connected to the PC running the Echo software, where the audio files will be stored and embedded thereafter in the call logging software.



#### **Features & benefits**

- USB 2.0 interface, USB powered
- Non intrusive and undetectable high impedance passive monitoring
- Dialled number and Caller ID signalling support
- Models for PRI-ISDN, BRI-ISDN /  $S_0$  bus, Digital handsets and Analog handset / trunk lines
- Digital Signal Processors for voice streaming and on-board D-channel protocol processing
- Fractional and 30 channel PRI-ISDN / E1 models
- · Supports 64kbit/s A-law and high quality compressed 36kbit/s speech encoding
- Analog models support DTMF, FSK Caller ID, AGC and speech detection
- Digital models support all major D-channel signalling protocols and highly accurate DigitalVox start/stop trigerring for non-protocol applications
- 1, 2, 4 and 8 channel Analog, Digital handset and S<sub>0</sub> bus models
- CE, FCC and RoHS compliance
- Supports Windows XP SP3, Vista, 7, 8, 10, Server 2003, 2008 and 2012

#### Inputs

Standard analogue line (POTS), basic rate (BRI) and primary rate (PRI) ISDN inputs are available, via the back panel which uses an RJ45 socket and a USB port.

#### **Dimensions**

SIZE: L 155mm x W 100mm x H 20mm. WEIGHT: 260g

#### **Power requirements**

The hardware device is connected to the host server by way of a high-speed USB2.0 cable from the rear of the unit.

### **PCI cards**

The Echo PCI Voice Recording boards are designed for medium to large organisation. Each card only occupies a single standard 5V PCI slot and combines a compact form factor with very low power consumption.



On-board DSPs and large streaming buffers impose very little demand on system and application resources.

D-channel signalling supports Call Setup, Connect, Clear, DDI number and CLI number decoding. It also supports all major ISDN variants, Q.SIG, DASS-2, DPNSS and various PBX Digital handset D-channel protocols. This means that simultaneous recording on all channels of both upstream and downstream sides of a conversation is possible.

The speech encoding/compression used is 64kbit/s A-law (G.711), 36kbit/s proprietary encoding. The estimated MTBF is 600,000 hours.

#### **Features & benefits**

- High density half length PCI card
- Fits in every standard 5V PCI slot
- · Non intrusive and undetectable high impedance passive monitoring
- Dialled number and Caller ID signalling support
- Digital Signal Processors for voice streaming and on-board D-channel protocol processing
- Models for PRI-ISDN, T1-ISDN, BRI-ISDN, Digital handsets and Analog handset / trunk lines
- 1, 2, 4, 8 and 16 channel Analog, Digital handset and 4-wire So bus models
- Fractional, 30 channel and 60 channel PRI-ISDN / E1 models
- 23 channel and 46 channel T1-ISDN models
- · Protocol support for all major ISDN variants, Q.SIG, DASS-2 and DPNSS
- Analog models support DTMF, FSK Caller ID, AGC and speech detection
- Digital models support all major PBX with highly accurate DigitalVox start/stop triggering and D-channel support for various PBXs
- Accurate start/stop triggering for non-protocol E1/T1 PCM applications
- Supports Windows XP SP3, Vista, 7, 8, 10, Server 2003, 2008 and 2012
- 64kbit/s A-law and high quality compressed 36kbit/s speech encoding
- · Supports writing of directly-playable WAV files to disk
- CE, FCC and RoHS compliance

#### **Technical specifications**

Mechanical characteristics	Half length PCI card
Operating temperature	0 °C to +60 °C
Humidity	5% to 80% non-condensing
Maximum power requirements	+5Vdc (400 mA), +12Vdc (50 mA), -12Vdc (50 mA)
Operating Systems supported	All 32-bit and 64-bit Windows editions
Maximum density	Up to 12 cards or 720 ports per chassis

#### **PRI specifications**

Rate	E1 (2.048Mb/s), T1 (1.544Mb/s)
AC Impedance	1100

Maximum tap length	100m (Terminated), 10m (Unterminated)
Protocol support	All major ISDN variants: Q.SIG, DASS-2, DPNSS

#### **BRI specifications**

Туре	4-wire S <sub>0</sub> bus
AC Impedance	Line matched
Maximum tap length	500m
Protocol support	Euro-ISDN

#### **Digital Extension Interface specification**

Туре	4-wire S <sub>0</sub> bus
AC Impedance	Line matched
Maximum tap length	500m
Protocol support	Euro-ISDN

### Array

The Echo Array is a very high density voice packetizer for building large scale PRI/E1 based TDM voice recording systems.



It includes embedded DSP arrays for D-channel decoding, voice processing and IP packet streaming.

The rugged design, low power consumption, redundant power supplies and TDM module hot-swap facility results in unmatched reliability.

D-channel signalling supports Call Setup, Connect, Clear, DDI number and CLI number decoding. It also supports all major ISDN variants, Q.SIG, DASS-2 and DPNSS.

It has highly secure encrypted IP streaming of the mixed upstream and downstream sides of a conversation.

The speech encoding/compression is 64kbit/s A-law (G.711), 36kbit/s proprietary encoding. The estimated MTBF is 600,000 hours.

#### **Features & benefits**

- 3U and 7U height 19" rackmount models with respectively 40 and 80 PRI / E1 interfaces (1200 and 2400 TDM channels)
- Non intrusive and undetectable high impedance passive monitoring
- Dialled number and Caller ID signalling support
- Digital Signal Processor Arrays for voice streaming and D-channel protocol processing
- Supports Windows XP SP3, Vista, 7, 8, 10, Server 2003, 2008 and 2012
- Remote TCP/IP accessibility for system configuration and FLASH memory upgrading
- Fully stand-alone embedded operation
- Redundant power supplies and full hot-swap facility for the plug-in modules
- 64kbit/s A-law and high quality compressed 36kbit/s speech encoding
- · Supports writing of directly playable WAV files to disk

- Highly secure encrypted IP streaming
- Custom IP output formats possible for supporting existing IP Recording systems
- CE, FCC and RoHS compliance

#### **Technical specifications**

Mechanical Characteristics	3U or 7U height 19" rack
Operating temperature	0 °C to +60 °C
Humidity	5% to 80% non-condensing
Maximum power requirements	+5Vdc (400 mA), +12Vdc (50 mA), -12Vdc (50 mA)
Operating Systems supported	All 32-bit and 64-bit Windows editions
Maximum density	Virtually unlimited, 2000+ ports total

#### **PRI specifications**

Rate	E1 (2.048Mb/s), T1 (1.544Mb/s)
AC Impedance	1100
Maximum tap length	100m (Terminated), 10m (Unterminated)
Protocol support	All major ISDN variants: Q.SIG, DASS-2, DPNSS

### SIP/VoIP call recording

### **Port mirroring**

The call recording of SIP lines requires a port mirroring setup, which enables the monitoring of network packets. The port mirroring allows the SIP traffic going through one port on your managed switch to be copied to another port, for packet analysis.

The image below illustrates the general configuration involved in port mirroring.



Your telephone system will be connected to one port on your managed switch, sourcing the SIP traffic, which will then be configured to send a copy of all SIP packets to another port, connected to one of the network cards on the Echo server.

A

For detailed information on how to configure port mirroring on your network switch, check the manufacturer documentation.

### SCCP/SIP extension-side configuration

Below is an example of a network layout for an installation that would allow recording of internal and external calls across the telephone network, depending on the configuration.



Monitoring handsets that are capable of either SCCP or SIP communication allow generic recording to occur, regardless of what combination of PSTN interfaces are in use, e.g. SIP, PRI, BRI, analogue.

To record extension-side audio, the monitoring can be configured in two ways. The first method will record external calls only, whereas the second will record all calls but will add further load to the managed switch.

Field	Description
External only	The managed switch needs to be configured to mirror the port where the telephone system connects to the telephone network.
All calls	The managed switch needs to be configured to mirror every port that has a telephone connected to it, but the telephone system's network connection doesn't need to be mirrored.
	This method allows the collection of the internal RTP traffic that would not be seen at the telephone system's network connection.

### **SIP trunk configuration**

Below is an example of a network layout for an installation that would allow recording of external calls across SIP trunks.



Monitoring SIP trunks allows any phone system or handsets to be recorded. This includes proprietary handsets which would normally be expensive when recording extension-side.

The managed switch needs to be configured to mirror the port where the telephone system would be sending the SIP/RTP call information.

### Echo setup

### **Installing Echo**

To install Echo, click on the setup package and follow the on-screen instructions.

🕎 Setup - Echo	
	Welcome to the Echo Setup Wizard
	This will install Echo v1.0.0.95 on your computer.
	It is recommended that you close any other setup programs before continuing.
	Click Next to continue, or Cancel to exit Setup.
	Next > Cancel

If you are monitoring ISDN channels, ensure the correct option is selected, according to your hardware device, and then, follow the on-screen instructions to complete the driver installation.

🗳 Setup - Echo		
Select Additional Tasks		
Which additional tasks should be performed	?	<u>کی</u>
Select the additional tasks you would like Se dick Next.	tup to perform while installing	Echo, then
Install USB Magic Box drivers		
☑ Install PCI card drivers		
	< Back Next >	Cancel

If you are using SIP lines, no option should be ticked.

### **Configuring Echo**

After installation, the Echo settings screen will appear, where you can configure the options detailed below:

### VOIP

VOIP Telecom Web Server Audio	ina
Adapter name	IP address
Buffer size 65535 bytes Protocol(s) UDP IV TCP End session if no audio after	SIP Port(s) 5060 Add Remove
Service status	SCCP port 2000

Field	Description
Choose a name	Enter a unique name to identify this installation; if you are installing more than one instance of Echo on your network, the name you enter here will be used in the subsequent setup of your TIM Plus or TIM Enterprise server.
Network adapter	Choose the adapter that will receive SIP/SCCP and media packets from your network.
Buffer size	The socket buffer value is the maximum size (in bytes) that you'd expect your network adapter to receive from your network. The default value of 65536 should be sufficient for most networks.
Protocols	The types of IP packets that you know Echo should look for when inspecting passing traffic on your network adapter. If you know for sure that your audio signal or media packets will never be TCP packets, you should deselect TCP to save CPU load; the less traffic that Echo needs to inspect, the more concurrent calls it can handle with the same hardware.

SIP port(s)	The UDP/TCP port number(s) that will be used for SIP signalling traffic on your network. Usually this is a single port (5060) but, in the case of media proxies or multiple SIP registrars on the same network segment, further ports may need to be entered here.
SCCP port	The TCP/UDP port that you expect Cisco SCCP (Skinny) packets to be transmitted on. This value is normally port 2000.
End session if no audio	This value specifies (in seconds) the maximum amount of time that Echo will continue monitoring a SIP or SCCP session in the event of no traffic, before it considers the session abandoned.

#### Web server

Choose a name to help you identify the location of this Echo installation My Echo Installation VOIP Telecom Web Server Audio
My Echo Installation
VOTP Telecom Web Server Audio
Web server settings
Port 8088
Username admin
Password *****
Service status
Apply
Service is running Stop Close

Setting	Description
Port	The port number that the internal web service that Echo should listen on
Username	The username required to access the Echo web service
Password	The password required to access the Echo web service

### Audio

Echo settings	
Choose a name to help you identify the location of this Echo installation	
My Echo Installation	
VOIP Telecom Web Server Audio	
Audio settings	
Quality 7 👘 10 = best	
Compress timeout 300 secs	
Select a base folder to store audio files	
	Change
Service status	Apply
Service status Service is running Stop	Apply

Setting	Description
Quality	The desired quality of audio recordings after compression. The higher the quality, the better the audio will sound, but the bigger the files on disk will be
Compress timeout	The amount of time (in seconds) that Echo will wait for an audio file to be compressed. By default, this is set on 2400 seconds (40 minutes).
Change	By default, the audio files are stored in the following location: C:\ProgramData\Tri-Line\Echo\audio. To select a different folder, click on the Change button and enter the preferred location.

When you have finished configuring Echo, click on the Apply button to save the settings to disk and then start the Echo service.

You can verify that the Echo service is running by connecting to its web service at: http://localhost:8088/. If you changed the Web server settings (above), the address may differ.

### **Echo interface**

8

To access the web interface of the Echo software, open a web browser and, in the address bar, type in the IP address or hostname of the PC running Echo, followed by the port number it's listening on (e.g. http://192.168.0.100:8088).

The main dashboard screen will display a list of calls, with the most recent at the top of the list. The live calls that are still in progress will be highlighted in yellow, as shown in the screenshot below:

Echo	× +				
€ € 192.168.0.1	00:8088		⊽ C Q Search	+ ☆ ★	â ♥ ⋪ ⊜ * · =
Echo	<b>)</b> 1.0.0.95				
A Home	Devices 🌣 Settings				
					Q Search C Refresh
	Started <b>V</b>	From	То	Duration	
ISDN	17:09:37	02072652600	189907792585808	00:00:0-37	0
ISDN	17:07:58	02072652600	189907792585808	00:00:10	0
ISDN	16:58:25	02072652600	08447881000	00:02:58	di 🖄
ISDN	16:49:23	1612745187	652697	00:00:42	.h ᆂ
ISDN	16:43:15	02072652600	1899001818674810	00:00:24	at 🖄
ISDN	16:31:34	02072652600	189901253501661	00:06:50	.h ᆂ
ISDN	16:27:38	8451255400	652600	00:03:30	.h 🖄 🗸

To see the waveform of a call, click on the **i**con, as shown below:

Echo	× +				
€ € 192.16	58.0.100:8088		▼ C Q Search	<b>↓ ☆ ★</b> €	■ ▼
i Ec	<b>ho</b> 1.0.0.95				
A Home	E Devices Setting	S			
					Q Search C Refresh
	Started V	From	То	Duration	
ISDN	17:09:37	02072652600	189907792585808	00:00:17	•
ISDN	17:07:58	02072652600	189907792585808	00:00:10	•
ISDN					.h ±
ISDN	16:49:23	1612745187	652697	00:00:42	ai ᆂ
ISDN	16:43:15	02072652600	1899001818674810	00:00:24	a ±
ISDN	16:31:34	02072652600	189901253501661	00:06:50	at ᆂ
ISDN	16:27:38	8451255400	652600	00:03:30	11 ±
•					

To download a call, right-click on the icon and then select the save link as option, as shown below:

Echo	× +							- 0	) ×
	2.168.0.100:8088		⊽ C Search		e 🛡		ø	** -	=
	DO 1.0.0.95	is							
				Duration		<b>Q</b> Sear	ch 🕻	C Refre	sh
ISDN	17:00:37	02072652600	180007702585808	00:00:17		(	5		
ISDN	17:07:58	02072652600	189907792585808	00:00:10		(	5		
ISDN	16:58:25	02072652600	08447881000	00:02:58		1	.h 🛨	1	
ISDN	16:49:23	1612745187	652697	00:00:42	Op	en Link i en Link i	n New <u>T</u> in New V	ab /indow	
ISDN	16:43:15	02072652600	1899001818674810	00:00:24	Op	en Link i	n New P	rivate W	indow
ISDN	16:31:34	02072652600	189901253501661	00:06:50	Bo	okmark T	This <u>L</u> ink		
ISDN	16:27:38	8451255400	652600	00:03:30	Sa	re This L e Lin <u>k</u> A	.inк .s		
					Co	py Link L	.oc <u>a</u> tion	Save"	
					Ins	pect Eler	ment (Q)		
					🗣 Ins	pect Eler	ment wit	h Firebu	9
					<b>,</b> ,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,				V

The application also allows you to search for a call, by entering the required call details in the pop-up window that opens when you click on the se arch button:

Echo	× +												- 0	X
♦ ♦ @ 192.168.0.100:80	188			⊽ C' Q,	Sean	ch	1	▶ ♠	<b>†</b>			ø	* -	≡
Echo 1.0.4 A Home ■ Device	Q Search									×	1			*
	Period	From number				Call reference							_	
	Today			*. *							A Searc	ch ;	Refres	h
Sta	<ul> <li>Yesterday</li> <li>Last 7 days</li> </ul>	To number				Channel range								
ISDN 17:0	<ul> <li>Last month</li> </ul>			-		0	to	127	\$		•	•		
ISDN 17:0	<ul> <li>This month</li> <li>This year</li> </ul>	Time range				Call direction					•	•		
ISDN 16:	<ul> <li>Last year</li> </ul>	00:00	to	23:59		Inbound Out	utbound					.lı 土		
16:4	<ul> <li>All dates</li> <li>Quarters</li> </ul>	Duration (secs)				Device						.u ±		
16:4	Custom	0	to	86400 🚔						<b>-</b>		. L .		
15DN 16.1												¥	-	
	limit number of results to 100													
		ι το σ	1	V   F	"	ምሳጥሳ	'	ľ	Sear	ch			•	

If you would like to see to which device the call recorder is connected to, such as ISDN, SIP or PCI card, click on the Devices tab.



If you need to apply or update your software license, click on the settings tab and then on the Show license button. The license details will be displayed in a new window, as shown below:

Echo	× +													×
♦ ♦ ④ 192.168.0.100.808	8					earch		÷	⋒	<b>†</b>   6		ø	* -	≡
<ul> <li>♦ € 192163.0.100.303</li> <li>♦ Home ■ Devis</li> <li>License</li> <li>Show license</li> </ul>	8 Software I Current licen: Expires Channels Echo can retr automatically Check now	icense se details 01/01/2016 10 ieve an updated lic if the server can co	ense from your vende nnnect to the internet.	r	v     C     Q, St   If you cannot with the production of the productin of the production of the producting of the producting of the pr	retrieve a license ar ict key below: RODUCT KEY xxEfylayfcLUV.BCe DUCT KEY e new license into th	utomatica rLsbAchE e followin	+ ally, pr b7/IO	vvide s uzalsb	+ E	ndor		98   <b>-</b>	
					Apply licens	e				Cid	.# ose			

### **Echo integration**

### **Overview**

TIM Talk is our integrated call logging and recording solution, combining one or more Echo installations with one of our call logging software, TIM Plus or Enterprise.

Unlike other call recorders that are separate and require special client software, our integrated solution enables all calls to be securely played through the existing web interface of the call logger, using a standard web browser.





### **Integrating with TIM Plus**

After installing Echo into your voice network, you need to configure its presence in TIM Plus.

Ensuring you are logged in to TIM Plus as an administrator, click on the Settings tab and select the voice recording option from the left-hand side menu. To add a voice recording option, click on the New tab, as shown below:



A new window will appear, allowing you to configure the settings of your call recording device.

Recording device	settings		X
General	Ch	annel map	
Device name			
Choose the type of rea	cording device:		
Туре	Magic Box - PF	स	-
Enter specific settings	for the selected	device:	
ID	0		
Host & Port	127.0.0.1		8088
Time offset	60	seconds	
	Ca	ncel	Add

Setting	Description
Device name	The name of your recording device
Туре	The type of telephone lines you are using: PRI, BRI, Analogue, VOIP
ID	The unique identifier of each call recording device
Host & Port	The IP address of the recording device, or computer to which the device is attached
Time offset	The time interval around which TIM Plus will search for calls when matching audio files

### Voice recording - overview video

### Integrating with TIM Enterprise

After installing Echo into your voice network, you need to configure its presence in TIM Enterprise, by following the steps below:

1. Log in to TIM Enterprise, drill-down to the Directory level where you want to add the call recording device and click on the New object t ab, as shown below:

timenterprise) Reports Directory Call View Ta	riff Editor Al	erts
🛧 \ UK \ London \		
💼 👌 🔍 📿 🚺 New object 17 items Clipboa	rd 0 items	
	aste 🌧 Clear	
8 All London		
C Accounts		
Customer Services		
🗋 Development		
Сп		
🗀 London Sales		
London E		
🖬 London Daily Stats		
🖬 London Weekly Stats		
🚍 Avaya IP Office		

2. In the new window that opens, select the Magic Box object from the Other object list, enter a name and select the type of telephone lines you are using (PRI, BRI, Analogue, VOIP), as shown below:

🍈 Add new object			
Organisation unit	Other object		
<ul> <li>Channel Group</li> <li>Cost Centre</li> <li>Division</li> <li>Group</li> <li>Reporting Collection</li> <li>Site</li> </ul>	<ul> <li>Alarm</li> <li>Channel</li> <li>PBX</li> <li>Magic Box</li> <li>User</li> <li>Display Board</li> <li>Question</li> </ul>	<ul> <li>Billing Charge</li> <li>Stats Collector</li> <li>LCR Plan</li> <li>Tariff Modifier</li> <li>Web User</li> <li>Directory Sync</li> </ul>	
An individual question a user	must answer in order to score	calls	
Enter a name for the object	PRI Magic Box		
Model	PRI (ISDN-30)	Ə Cancel 📙 Add	<u>d</u>

3. A new window will open, allowing you to configure the properties of your recording device object.

🚙 PRI Magic Box					
General PRT TN	PRI Magic Box	[			
PRI OUT	General setting	r poor generation		a Ar ga a a	
	Name Box ID IP address:Port Time tolerance	New Magic Box 0 127.0.0.1 60	8088	seconds	
				Cancel	📕 Save

Setting	Description
Name	The name of your recording device
Box ID	The unique identifier of each call recording device

IP address: Port	The IP address of the recording device, or computer to which the device is attached and the port number
Time tolerance	The time interval around which TIM Enterprise will search for calls when matching audio files

### Voice recording - overview video

### Acessing your call recordings

Call recordings are accessible via the web interface of your call logging solution. To listen or save a call recording, click on the 🖓 icon in the Call view screen or any itemised call logging report, as shown below:

Firefox <b>*</b>	ise 3.0.0.87 - Yo	ur Company .	. +										x
← ⇒	enterprise. <b>ca</b>	ll-logger.com					<b>☆</b> ⊽ C <sup>4</sup>	<mark>8</mark> <del>-</del> Ga	ogle	٩	Ŧ	俞	
timent	erprise				Repo	orts Direc	tory	Call Vie	w Tariff E	ditor		Alert	5
Call Vi	iew								🕨 Clear all 🛛 🎾	Head	ers		
Time v	Response					CLI	Dialled n						
17:23:16	0	00:03:33	0.00	09 October 2013	Holly Day	02074378594			London		Q	습 다 다	۲
17:23:01	0	00:02:42	0.00	09 October 2013	Doug Hole				120025		Q		۲
17:22:58	0	00:02:51	0.00	09 October 2013	Jack Wilson		2009		Earl E. Riser				۲
17:22:53	0	00:00:32	0.00	09 October 2013	Hazel Nutt	02089623450			London		$\overline{\mathbf{v}}$		۲
17:22:16	20	00:02:23	0.00	09 October 2013	Dr. Payne	02071829268			London		$\overline{\mathbf{v}}$		۲
17:20:58	0	00:00:04	0.00	09 October 2013	Norman Foster		2084		Sarah Kokova				۲
17:19:33	0	00:00:33	0.00	09 October 2013	Mike Francis	07817222380			Orange		$\overline{\mathbf{v}}$		۲
17:18:38	0	00:00:18	0.00	09 October 2013	Barb E. Dahl	02073002904			London		Q		$\oplus$
17:17:58	0	00:02:39	0.00	09 October 2013	Hazel Nutt	02078815600			London		$\nabla$		۲
17:17:06	0	00:00:51	0.00	09 October 2013	Rick Shaw				121003		5		nlav rec
17:16:38	0	00:01:14	0.00	09 October 2013	Dan Druff	07809765899			02		¢		() ()
17:16:13	0	00:04:46	0.00	09 October 2013	Barry Cade				120026		Q		۲
17:15:58	0	00:01:28	0.00	09 October 2013	Holly Day	02074940747			London		$\bigtriangledown$		۲
17:15:41	0	00:03:38	0.00	09 October 2013	Holly Wood				120035		Q		۲
17:15:31	0	00:04:08	0.00	09 October 2013	Roman Holiday				120014		Q		$\odot$
17:15:28	0	00:05:20	0.69	09 October 2013	Hutchison 3G		07723047	504	Bill Loney		Q		۲
17:14:58	0	00:01:02	0.00	09 October 2013	Mel Loewe	02070531000			London		$\overline{\mathbf{v}}$		۲
17:14:31	0	00:01:27	0.19	09 October 2013	T-Mobile		07908273	602	Kathy Burke		Q		$\oplus$
17:14:28	0	00:00:04	0.03	09 October 2013	London		02070010	000	Sandy Shore		Q		۲
17:14:18	0	00:00:32	0.00	09 October 2013	Mel Loewe	02076353629			London		Q		۲
17:14:08	0	00:00:47	0.10	09 October 2013	T-Mobile		07956836	710	Barry Cade		Q	∽ ri	$\langle \hat{+} \rangle$
09/10/2013	17:23:19 L	ogged in a	s 'Joe	Bloggs'									

A new window will open, allowing you to listen or save the call recording, as shown below:

Firefox       TIM Enterprise 3.0.0.87 - Your Company				- 0	X
♦ → @ enterprise.call-logger.com	8 - Googl	e 🔎	₽	⋒	
timenterprise Reports Directory	Call View	Tariff Edito	r	Alert	5
Call View	(	Clear all 🛛 🔀 He	aders		Save
Time <b>v</b> Response Duration Cost Date Destination CLI Dialled n					
17:20:58 0 00:00:04 0.00 09 October 2013 Norman Foster 2084	Sa	arah Kokova	Q	슈디	۲
I Inbound call from 02074940747 to Holly Day - 09 October 2013 17:15:58	× o	range	Ģ		•
Audio Audit trail Scoring Notes Polated calls Call info	Lo	ndon	Q		•
Autio Autin Sconing Holes Related cans can into	Lo	ondon	$\bigcirc$		•
	12	21003	Q		۲
	0	2	Ģ		۲
	12	20026	Ģ		۲
	Lo	ondon	Q		۲
Loading	12	20035	Q		۲
	12	20014	$\bigtriangledown$		۲
	Ві	ll Loney	$\bigcirc$		٠
	Lo	ondon	$\bigcirc$		۲
	Ка	athy Burke	Q		۲
🗧 Save recording	g Si	andy Shore	Q	<u>~</u> 1	۲
	Open	Link in New <u>T</u> ab			۲
	Open	Link in New Window		1	۲
	Open	LINK IN New Private	vindow	-11	۲
17:12:58 0 00:00:51 0.00 09 October 2013 Warren Pease 02073682480	Book	mark This <u>L</u> ink		_1	۲
17:12:46 0 00:00:24 0.00 09 October 2013 Neil Down	Savel	Lin <u>k</u> As		-1	۲
17:12:16 0 00:05:26 0.00 09 October 2013 Barry Cade	Copy This F	rame		, 🖆	۲
17:11:58 0 00:00:29 0.06 09 October 2013 Hutchison 3G 077377750				- 1	$(\hat{+})$
09/10/2013 17:21:32 Logged in as 'Joe Bloggs'	Inspe	ct Element (Q)			

## Call obfuscation (PCI-DSS)

TIM Plus/Enterprise offers the ability to obfuscate (mask out) one or more sections of the audio of a telephone call with an audible tone, preventing the listener from hearing the original speech, on playback

This is normally required for compliance in certain industries where regulations dictate that certain spoken information be masked out, e.g. the Payment Card Industry - Data Security Standard (PCI-DSS).

In this example, we will adopt the PCI-DSS example where telephone calls that contain spoken credit card information needs to be masked out by an audible tone, but only during those parts of the call when the card details are being spoken, leaving intact the rest of the call audio.

Our recommended option is Programmatic obfuscation. For this, we'll assume that agents (employees that make or receive telephone calls) utilise an in-house or third-party data entry system into which credit card detailed are entered using a computer.

### **Programmatic obfuscation**

Throughout this guide, we will adopt the PCI-DSS example above, where telephone calls that contain spoken credit card information need to be masked out by an audible tone, but only during those parts of the call when the card details are being spoken, leaving intact the rest of the call audio.

In this scenario, we'll assume that agents (employees that make or receive telephone calls) utilise an in-house or third-party data entry system into which credit card detailed are entered using a computer.

#### How it works

Considering TIM Plus or TIM Enterprise (in conjunction with one or more Magic boxes) records the call audio at strategic boundaries in your telecom infrastructure - usually your organisation's telephone lines, rather than each user's telephone handset - some reconciliation is normally required between those boundaries and the actual agent that handled the call.

By default, this reconciliation occurs automatically in TIM, which is how the agent-centric calls that you see in call reports are able to be associated (matched) with each call, as seen from the point of view of a telephone line which delivers calls to many agents.

During obfuscation, it is necessary that a user or device sends at least two signals to TIM. Together, these two signals allow our software to mask out the audio between the two points in time that each signal was received.



At the point in time during an agent's call when obfuscation is necessary - e.g. "Can I have your CVV number please?" is spoken by the agent - a signal is sent by the agent to TIM, which records the event along with the exact time it was sent. Similarly, when the sensitive part of the call is finished, a further signal is sent, which is also being recorded.

A single telephone call can contain more than one obfuscation and the number of signals required is always twice the amount of obfuscations in a call.

#### Assumptions

This guide assumes the following statements are true:

- · You have a licensed copy of TIM Plus or TIM Enterprise that includes voice recording
- Your installation is at least version 3.0.0.55

#### **Common solutions**

Taking the example of masking out some digits of a phone call when a credit card number is being quoted, most solution providers modify the data entry system that an agent uses.

#### Implementation

#### **HTTP request**

To send a start or stop signal, a simple HTTP GET request must be sent to the TIM web server.

Every request to the web server requires authentication, so ensure that the relevant HTTP authentication headers are sent with your request and that the username and password combination match an existing web user object in the Directory.

The response status code will indicate success or failure.

#### **Request format**

The request should be a GET request and take the following URL- encoded parameters, as per the following example:

http://192.168.0.1/signal.js?cmd=set&cat=4&type=1&objtype....

Valid parameters are described in the table below:

Parameter	Description
cat	Signal category. For audio masking, this value is always 0x04
type	The type of signal. Valid values for 0x04-category signals are:
	• 0x01 Mute On
	• 0x02 Mute Off
objtype	The type of object that this signal relates to. This can be one of two values:
	• user (a user object)
	• channel (a channel object)
objid	The unique ID of the object type as specified by the objtype parameter (above). This is used to locate the object in the Directory
	The region of the Directory to search in is specified by the key parameter (below) and governed by the access implied by the placement of the web user whose credentials are used to effect the web request
key	Specifies the key relating to a container object in the directory (or blank, implying the whole directory) whereby a search on the object specified by objtype and objid is performed below

Return values are specified as HTTP response status code. Although the body of some responses may contain informational text, you should not rely on this text to make any decisions as to whether the request was successful or not.

Valid status codes are as follows:

Parameter	Description
200	The signal was received and stored successfully

400	The request was not acceptable for one of the following reasons:
	<ul> <li>An invalid type parameter was specified. The type parameter is specific to the category specified by the cat parameter. Further, the type value (e.g. 0x01) can be used in multiple categories</li> <li>The objid was missing. Specify the ID of the object you want the signal to relate to</li> <li>The cat and type parameters - category and signal type, respectively - must be specified and cannot be zero</li> <li>The version of TIM Plus or TIM Enterprise you are running does not understand the signal.js script</li> </ul>
404	The object specified by the combination of the <b>objtype</b> and <b>objid</b> parameters - and optionally the <b>key</b> parameter - could not be found
500	Internal Server Error prevented the signal from being stored successfully. This may be due to a badly-configured database, or the lack of a signals table in the TIM Plus or Enterprise database

### **DTMF** obfuscation

The system permits manual obfuscation of calls using a DTMF key sequence at the user's handset. When a user enters a specified sequence of digits, the obfuscation begins, and any audio will be immediately and irreversibly overwritten with the tone specified in the /audio/obfuscatio n/tones/ section of the settings.xml file. To stop obfuscation, the stop key sequence must be entered, whereupon the audio will be written normally to the call's stream.

The start and stop key sequences can be defined in the settings.xml file located in {app}\ProgramData\Tri-Line\Echo, using the d tmfStart and dtmfStop elements. The following settings.xml snippet is an example of a configuration to enable manual call obfuscation when a user enters \*\*

```
<audio>
<obfuscation>
<dtmfStart>**</dtmfStart>
<dtmfStop>##</dtmfStop>
<dtmfTimeout>2000</dtmfTimeout>
<dtmfTimeout>50000</stopTimeout>
<toneFile>
<default>{app}/tones/alaw.wav</default>
<codec id="0">{app}/tones/ulaw.wav</default>
<codec id="8">{app}/tones/ulaw.wav</codec>
</toneFile>
</obfuscation>
</audio>
```

The dtmfTimeout element was provided to prevent a situation where extra digits have been erroneously entered, and therefore making it impossible to enter the correct sequence. After the specified DTMF timeout has passed, all the digits previously entered will be cleared. This also ensures that obfuscation does not occur during normal negotiation of DTMF telephone selection menus (inbound or outbound). To set a DTMF entry timeout, specify the value (in milliseconds) using the dtmfTimeout element.

Additionally, an obfuscation timeout can be specified to automatically stop the obfuscation after a specified period of time, to prevent the entire remaining call from being obfuscated in case the user forgets to manually stop the obfuscation. This timeout period can be specified (in milliseconds) using the stopTimeout element.



#### **Custom tones**

Custom tones can be added to the call recorder in order to obfuscate part of a call recording. Since the tones are raw byte data that replace portions of the original WAV audio, they must respect the codec of the original file, e.g. a-law, u-law, etc., and use the same sample size and rate, e.g. 16-bit, 8 KHz.

Once the files have been encoded accordingly, they should be copied to the following location: {app}\ProgramData\Tri-Line\Echo\tones.

To enable the system to use these custom tones during the obfuscation, open the settings.xml file, located in {app}\PrgramData\Tri-Li ne\Echo and add the following snippet, specifying the filename of the custom tone using the codec element. The id attribute determines the codec type, e.g. "0" = u-law, "8" = a-law, as per RFC3551 (page 32).

```
------
```

```
<audio>

<audio>

<obfuscation>

<toneFile>

<default>{app}/tones/alaw.wav</default>

<codec id="0">{app}/tones/ulaw.wav</codec>
<codec id="8">{app}/tones/alaw.wav</codec>
</codec id="8">{app}/tones/alaw.wav</codec>
</codec id="8">{app}/tones/alaw.wav</codec>
</obfuscation>
```

A <default> element can be added as the fall-back tone to use.

### **Call obfuscator**

A

To obtain a copy of the Call Obfuscator application, contact our Technical Support team.

To install and configure the Call Obfuscator application, follow the steps below:

1. Double-click on the setup package and follow the wizard in order to complete the installation.

😸 Call Obfuscator Setup	
	Welcome to the Call Obfuscator Setup Wizard
	The Setup Wizard allows you to change the way Call Obfuscator features are installed on your computer or to remove it from your computer. Click Next to continue or Cancel to exit the Setup Wizard.
	Back Next Cancel

2. Start the application from the All programs list and notice the 🛡 icon appearing in the system tray.



• Right-click on the icon and select Settings. A new window will open allowing you to configure the application.

	×
Port 80	Save Cancel
	1
	Port 80

Field	Description
IP address	The IP address of the PC running TIM Plus

Po	rt	The port number that TIM Plus is listening on	
Ex	tension	ID The extension number of the user whose calls should be obfuscated by this instance of the Call Obfuscator	
	0	A web user with the following credentials must exist in the call logger, in order to authenticate the http request set the application:	nt by
		username: admin	
		password: tntnet	

1. During a phone call, start the obfuscation by left-clicking on the vicon which, after some milliseconds, will change to vicon, indicating that the call is currently being obfuscated.



- 2. To stop an obfuscation, click on the 💛 icon, which will revert to 🔍, indicating that the obfuscation has ceased.
- 3. To exit the application, right-click on the icon and select Exit.

Multiple obfuscations can be performed on a single call by repeatedly clicking the start and stop icon.