ECHO

Advanced SIP recording
for small to mid-size organisations

Product Documentation

Last Updated: February 2016

Table of Contents

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

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_0 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 S_0 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 $^{\circ}$C to +60 $^{\circ}$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

Type	4-wire S_0 bus
AC Impedance	Line matched
Maximum tap length	500m
Protocol support	Euro-ISDN

Digital Extension Interface specification

Type	4-wire S_0 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 ^{o}C to +60 ^{o}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.
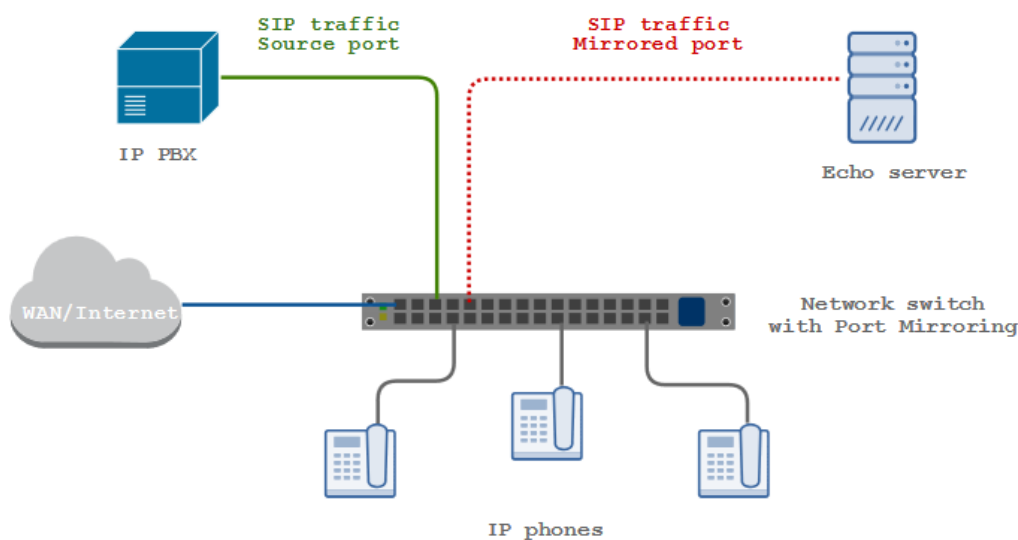
Your telephone system will be connected to one port on your managed switch, sourcing the SIP traffic, which will then be configured to send a copy of all SIP packets to another port, connected to one of the network cards on the Echo server.

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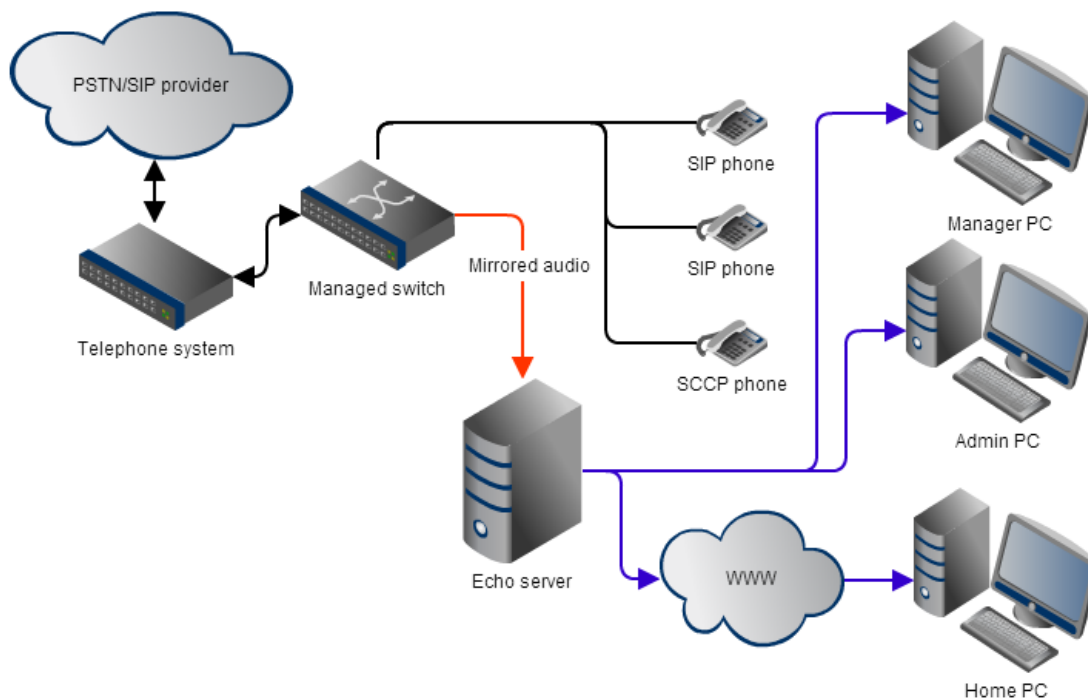
Monitoring handsets that are capable of either SCCP or SIP communication allow generic recording to occur, regardless of what combination of PSTN interfaces are in use, e.g. SIP, PRI, BRI, analogue.

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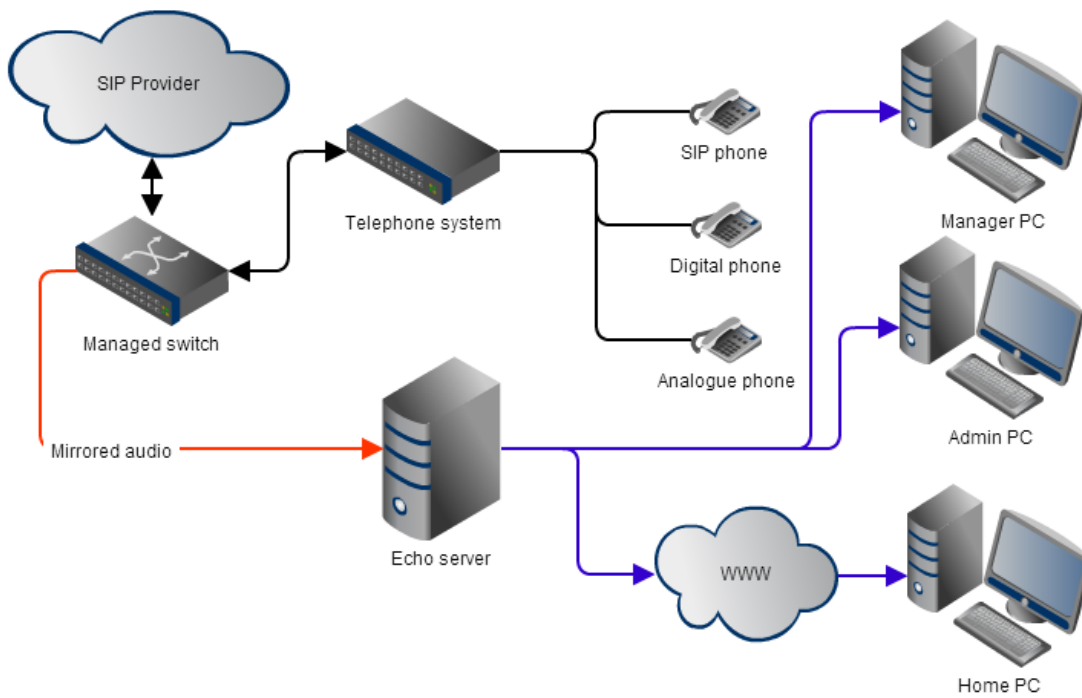
Monitoring SIP trunks allows any phone system or handsets to be recorded. This includes proprietary handsets which would normally be expensive when recording extension-side.

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.
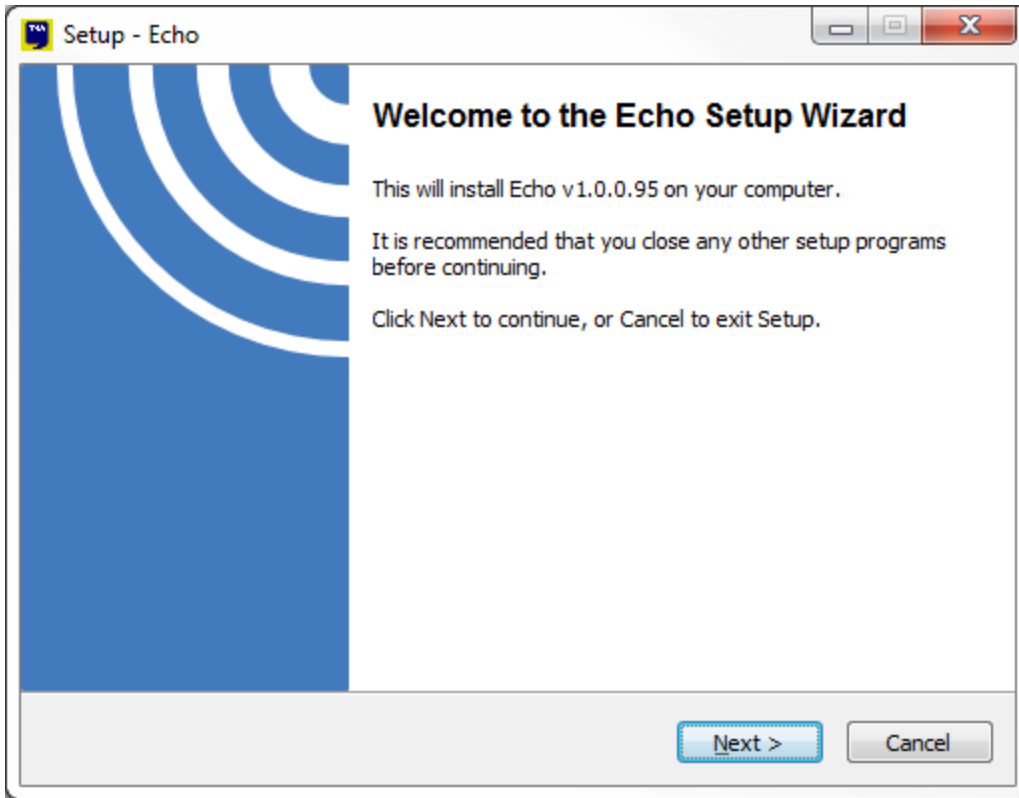
If you are monitoring ISDN channels, ensure the correct option is selected, according to your hardware device, and then, follow the on-screen instructions to complete the driver installation.
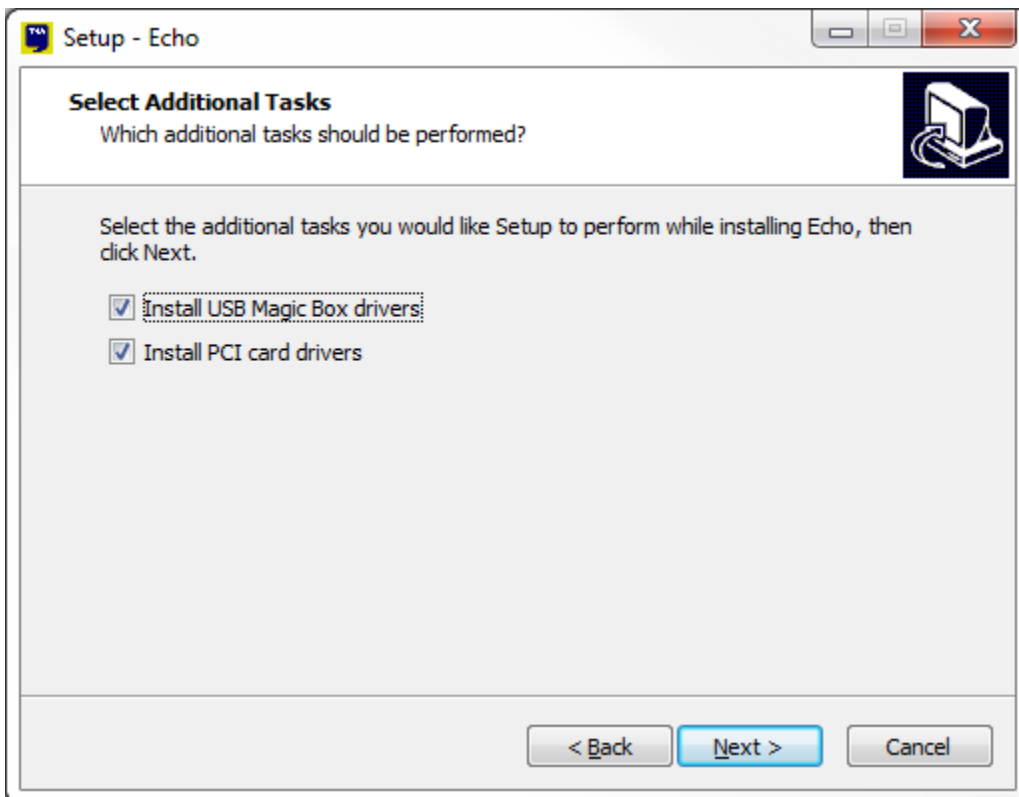
> ⓘ If you are using SIP lines, no option should be ticked.

Configuring Echo

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

VOIP

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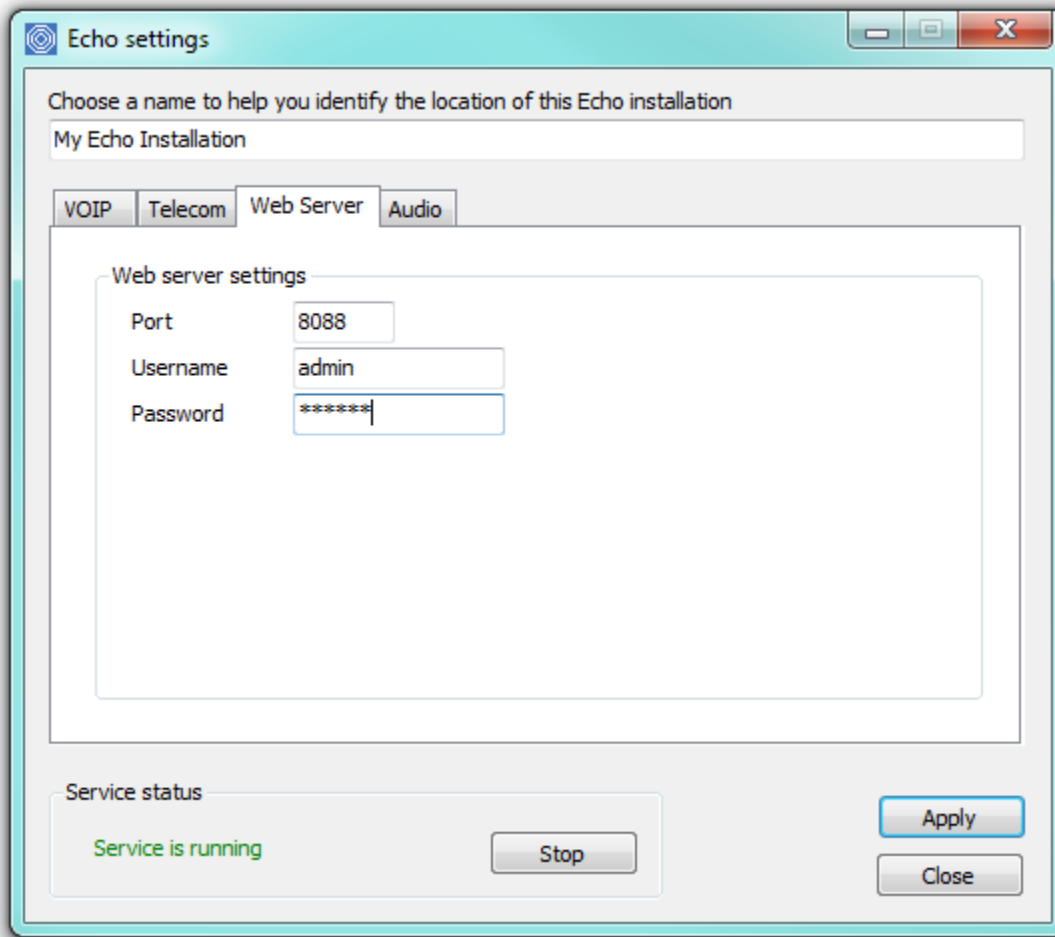
Setting	Description
Port	The port number that the internal web service that Echo should listen on
Username	The username required to access the Echo web service
Password	The password required to access the Echo web service

Audio
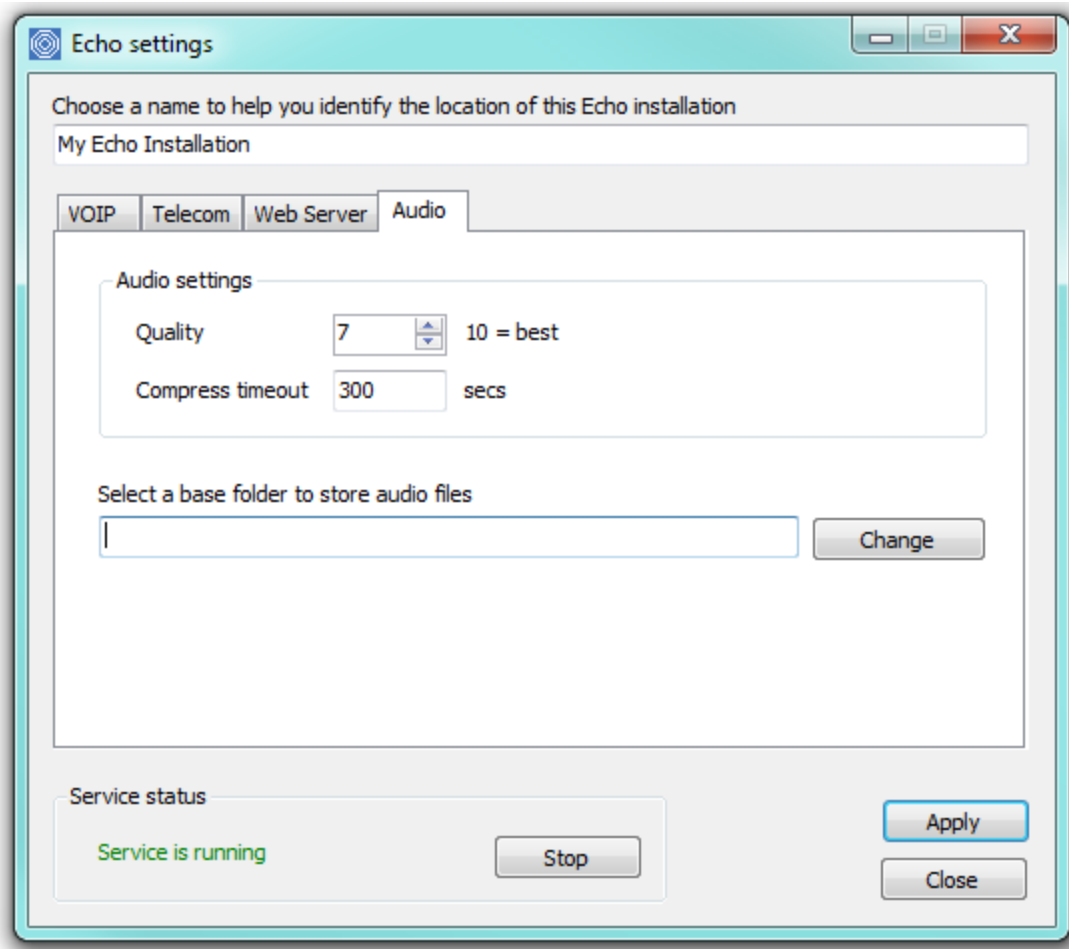
Setting	Description
Quality	The desired quality of audio recordings after compression. The higher the quality, the better the audio will sound, but the bigger the files on disk will be
Compress timeout	The amount of time (in seconds) that Echo will wait for an audio file to be compressed. By default, 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

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:
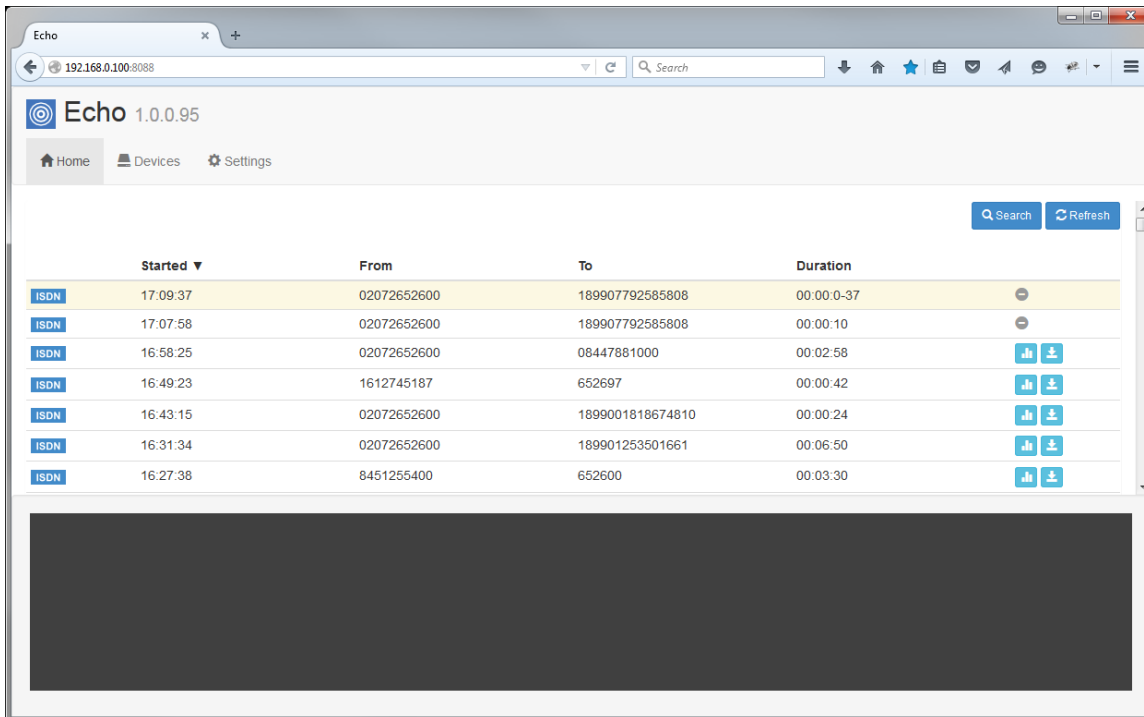
To see the `waveform` of a call, click on the [icon] icon, as shown below:
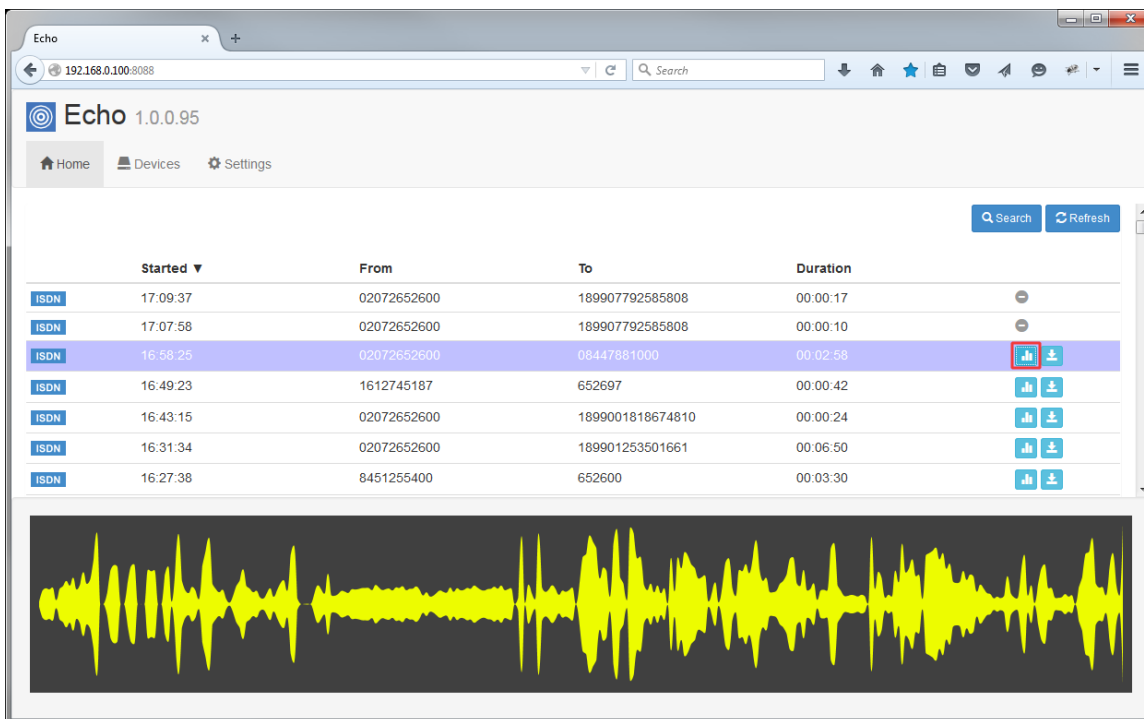

To download a call, right-click on the [icon] icon and then select the `Save link as` option, as shown below:
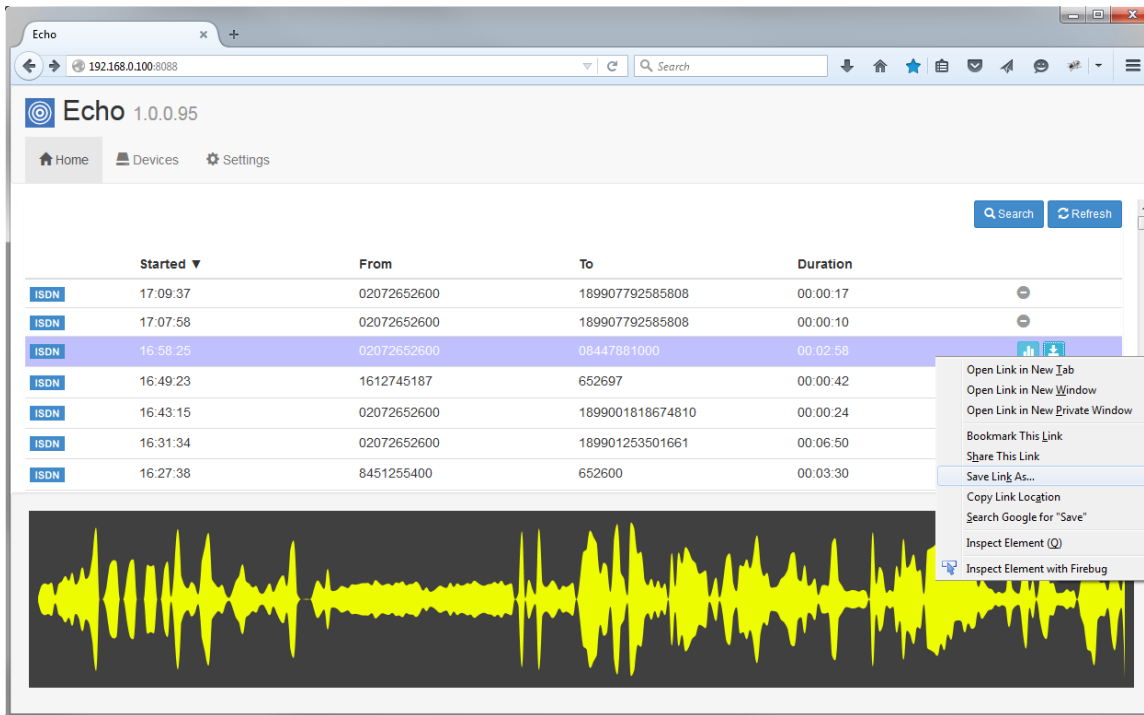
The application also allows you to search for a call, by entering the required call details in the pop-up window that opens when you click on the `Search` button:
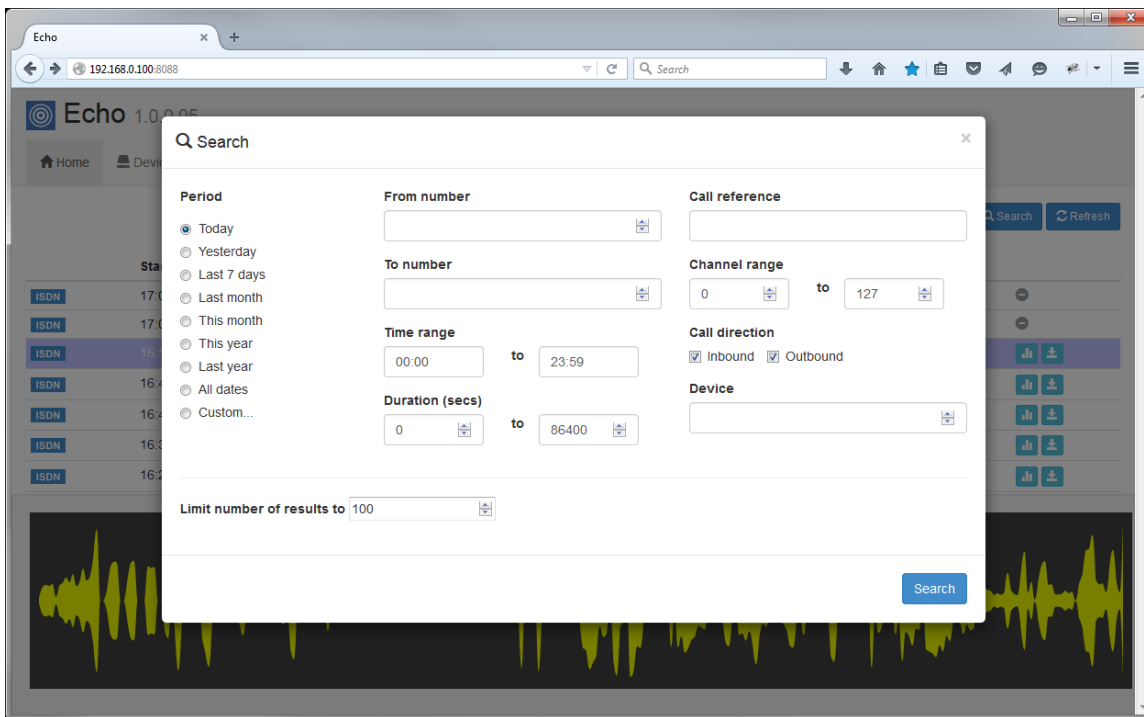
If you would like to see to which device the call recorder is connected to, such as ISDN, SIP or PCI card, click on the `Devices` tab.
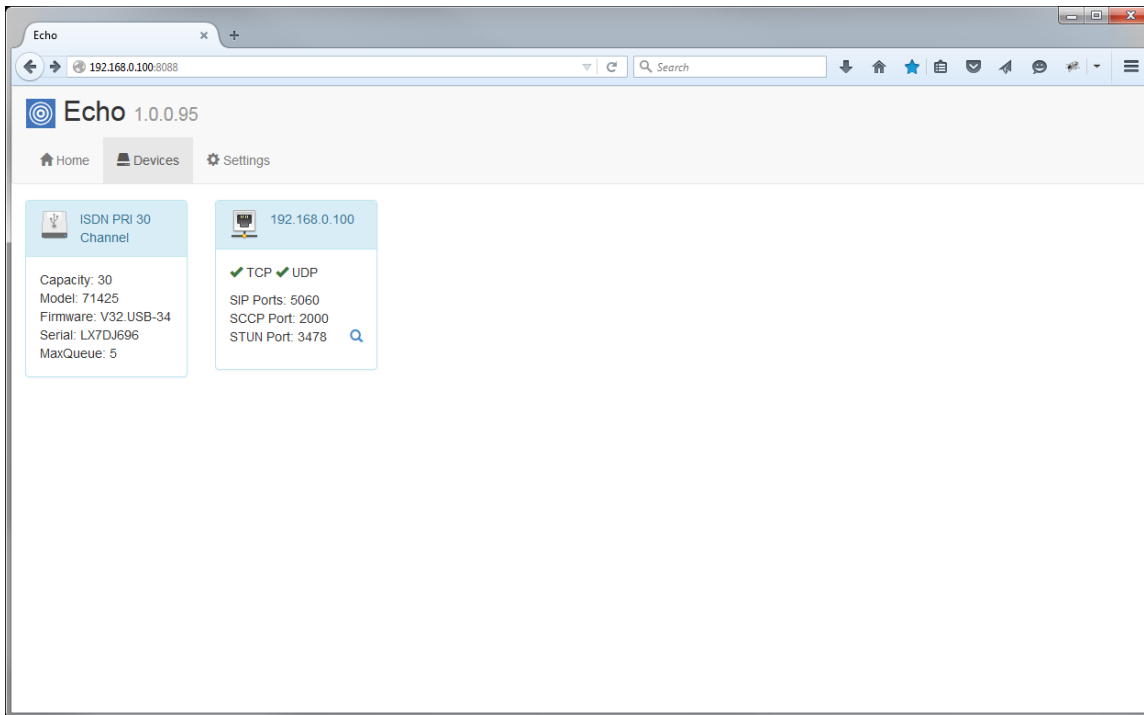
If you need to apply or update your software license, click on the `Settings` tab and then on the [Show license] button. The license details will be displayed in a new window, as shown below:
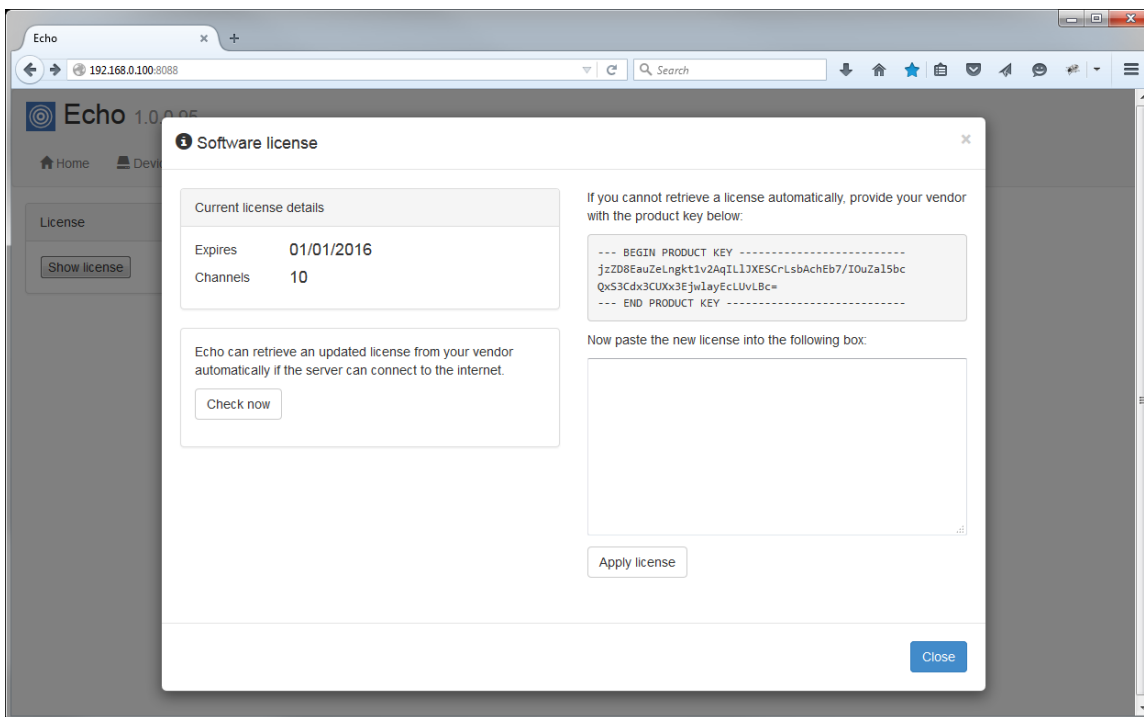
Echo integration

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.
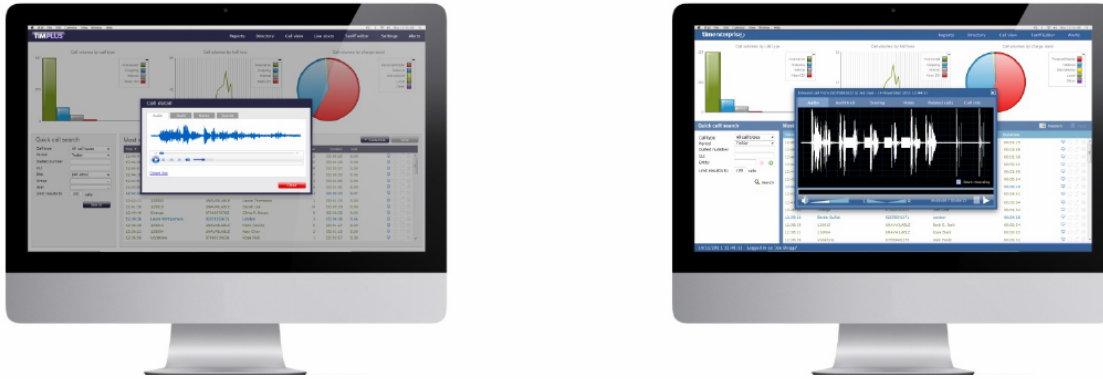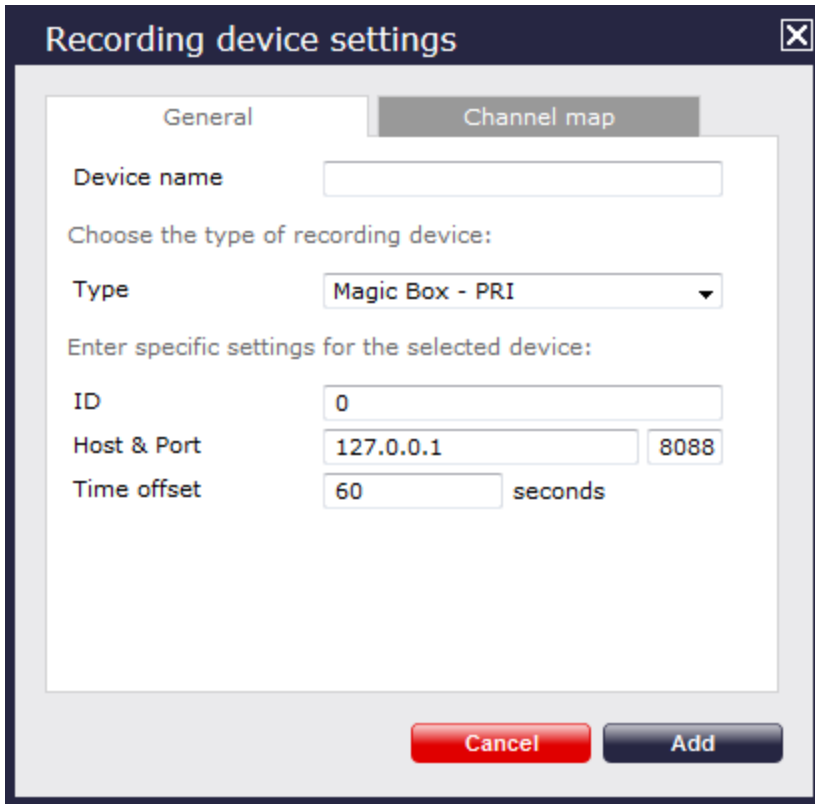
Integrating with TIM Plus

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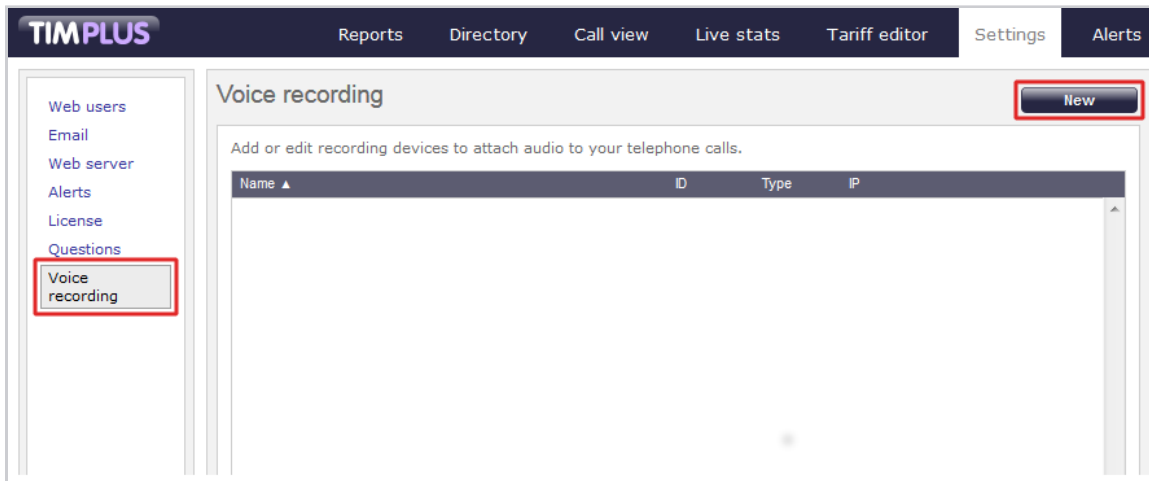
A new window will appear, allowing you to configure the settings of your call recording device.

| Setting	Description
Device name	The name of your recording device
Type	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
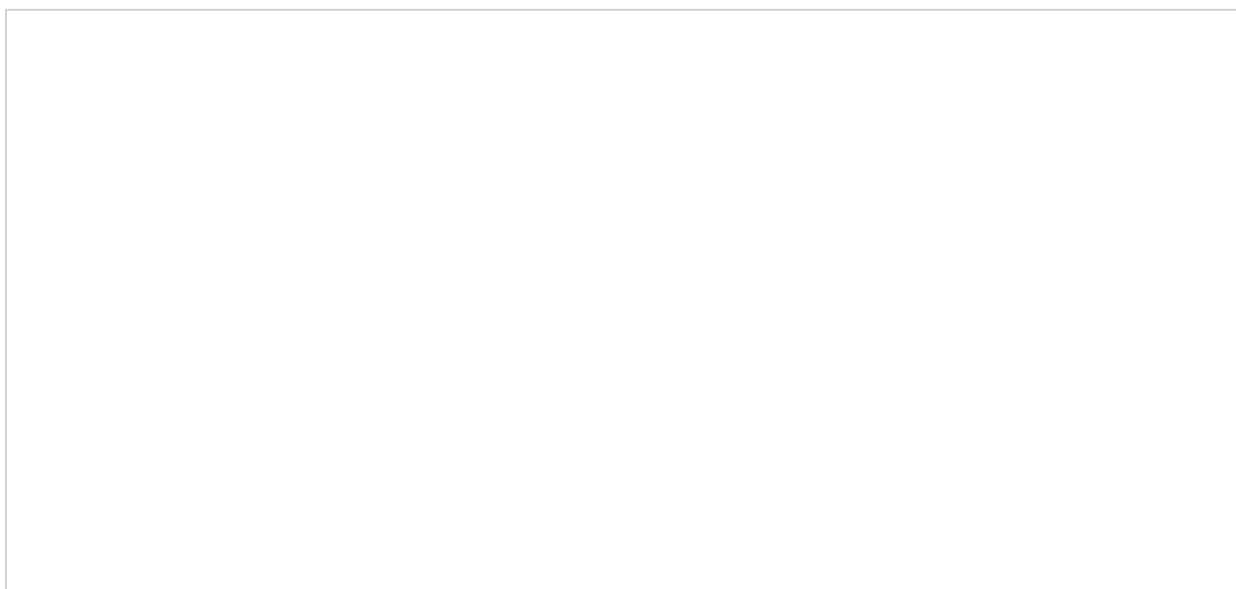
Integrating with TIM Enterprise

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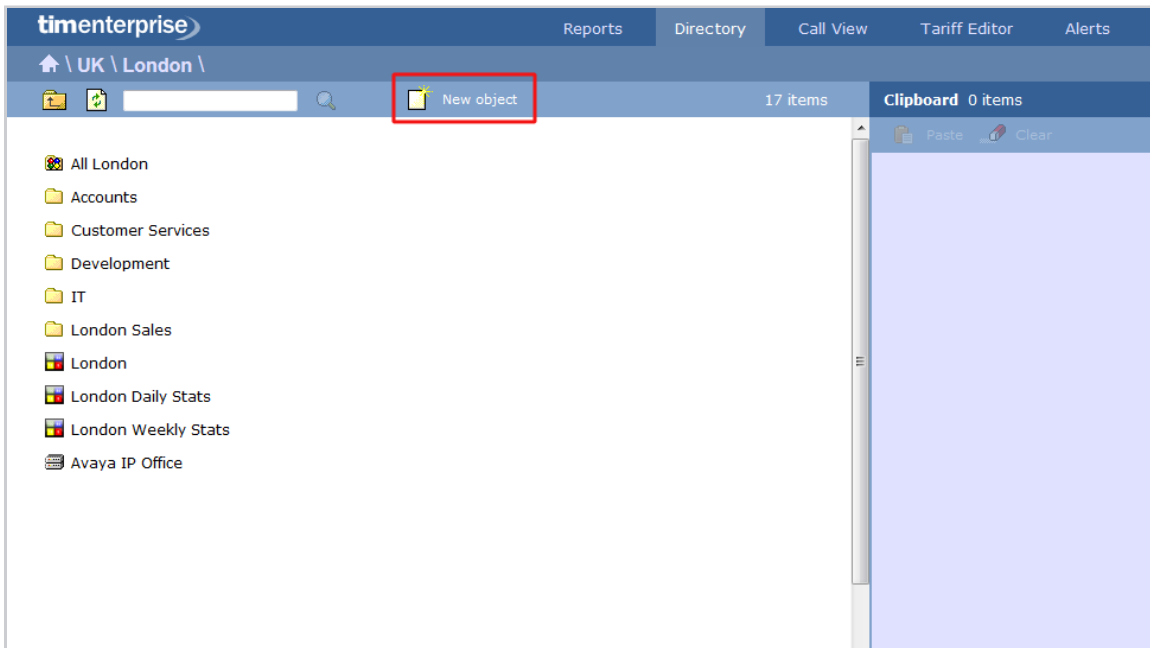
2. In the new window that opens, select the `Magic Box` object from the `Other object` list, enter a name and select the type of telephone lines you are using (PRI, BRI, Analogue, VOIP), as shown below:

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

| 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

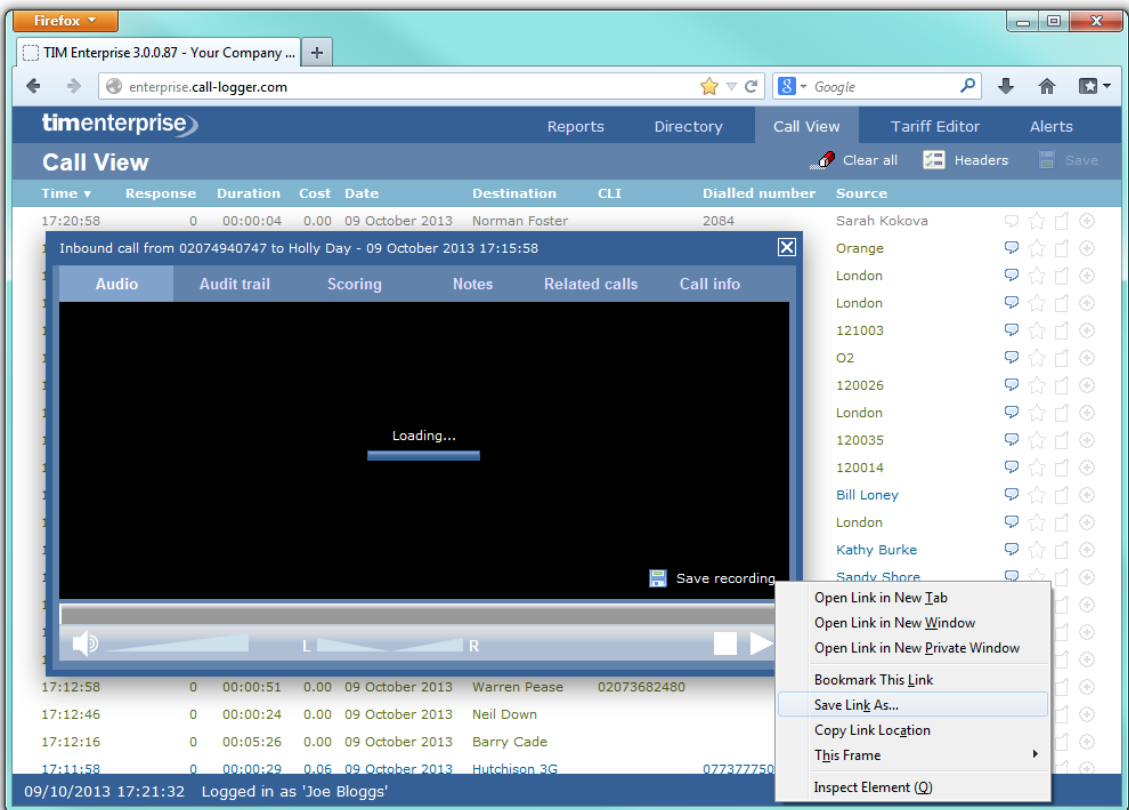## Acessing your call recordings

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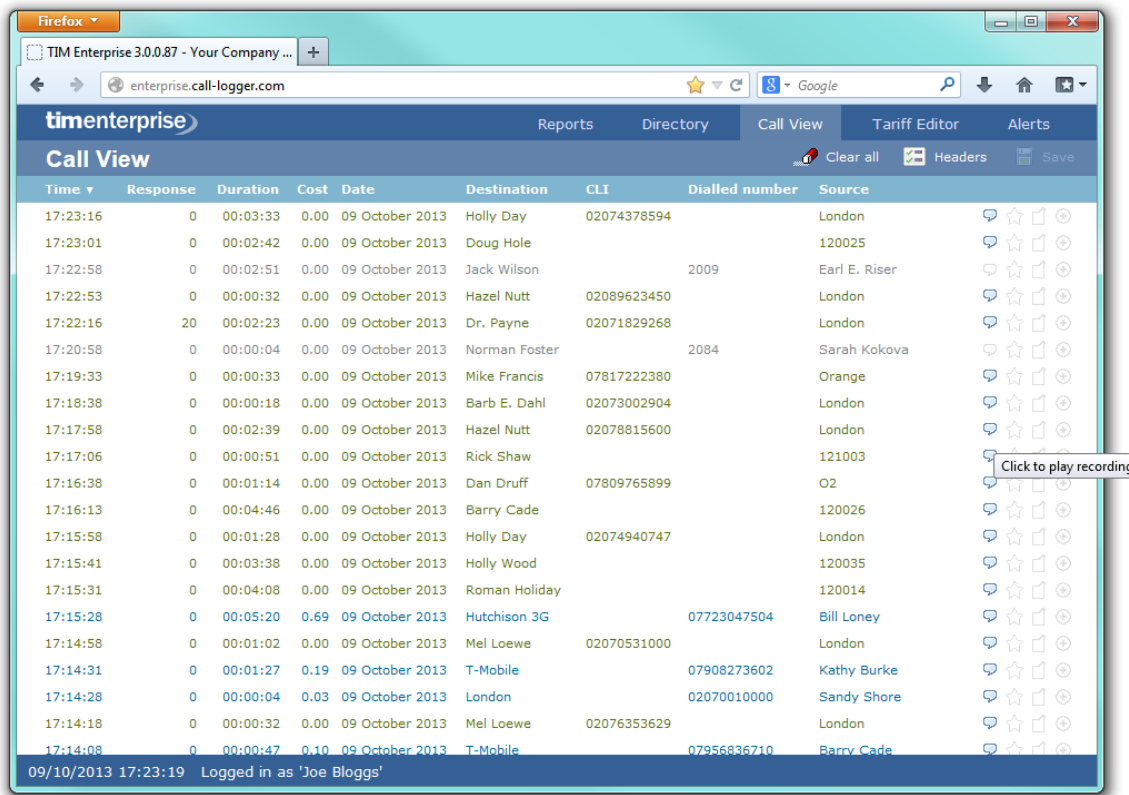
A new window will open, allowing you to listen or save the call recording, as shown below:

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.
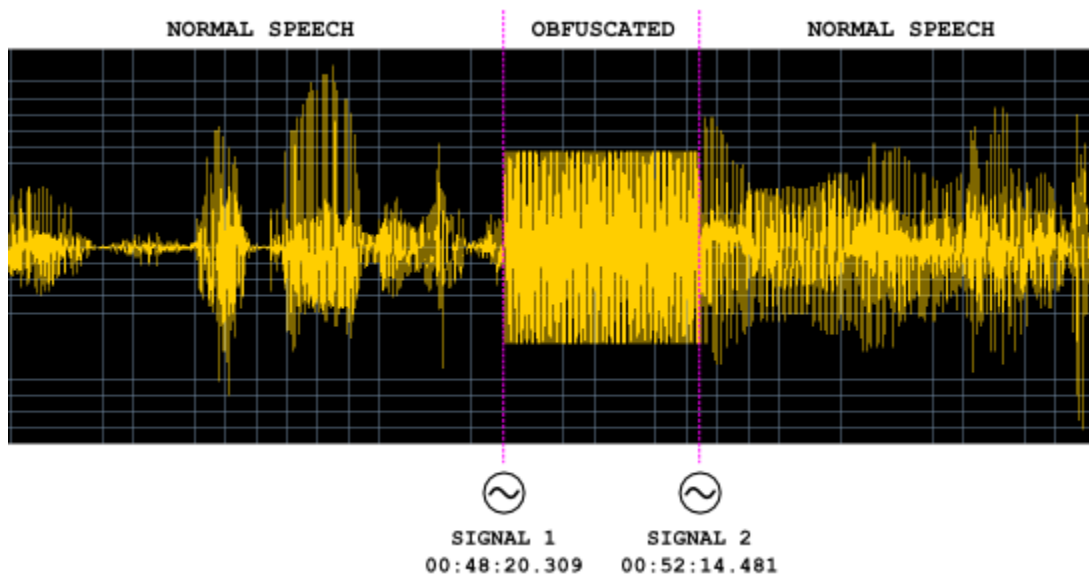
At the point in time during an agent's call when obfuscation is necessary - e.g. *"Can I have your CVV number please?"* is spoken by the agent - a signal is sent by the agent to TIM, which records the event along with the exact time it was sent. Similarly, when the sensitive part of the call is finished, 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:
	- An invalid `type` parameter was specified. The `type` parameter is specific to the category specified by the `cat` paramet er. Further, the `type` value (e.g. 0x01) can be used in multiple categories - The `objid` was missing. Specify the ID of the object you want the signal to relate to - The `cat` and `type` parameters - category and signal type, respectively - must be specified and cannot be zero - The version of TIM Plus or TIM Enterprise you are running does not understand the `signal.js` script
404	The object specified by the combination of the `objtype` and `objid` parameters - and optionally the `key` 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>
        <stopTimeout>50000</stopTimeout>
        <toneFile>
            <default>{app}/tones/alaw.wav</default>
            <codec id="0">{app}/tones/ulaw.wav</codec>
            <codec id="8">{app}/tones/alaw.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.

> ℹ️ For the changes to take effect, access the Windows Services and restart the Echo service.

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-Line\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>
    <obfuscation>
        <toneFile>
            <default>{app}/tones/alaw.wav</default>
            <codec id="0">{app}/tones/ulaw.wav</codec>
            <codec id="8">{app}/tones/alaw.wav</codec>
        </toneFile>
    </obfuscation>
</audio>
```
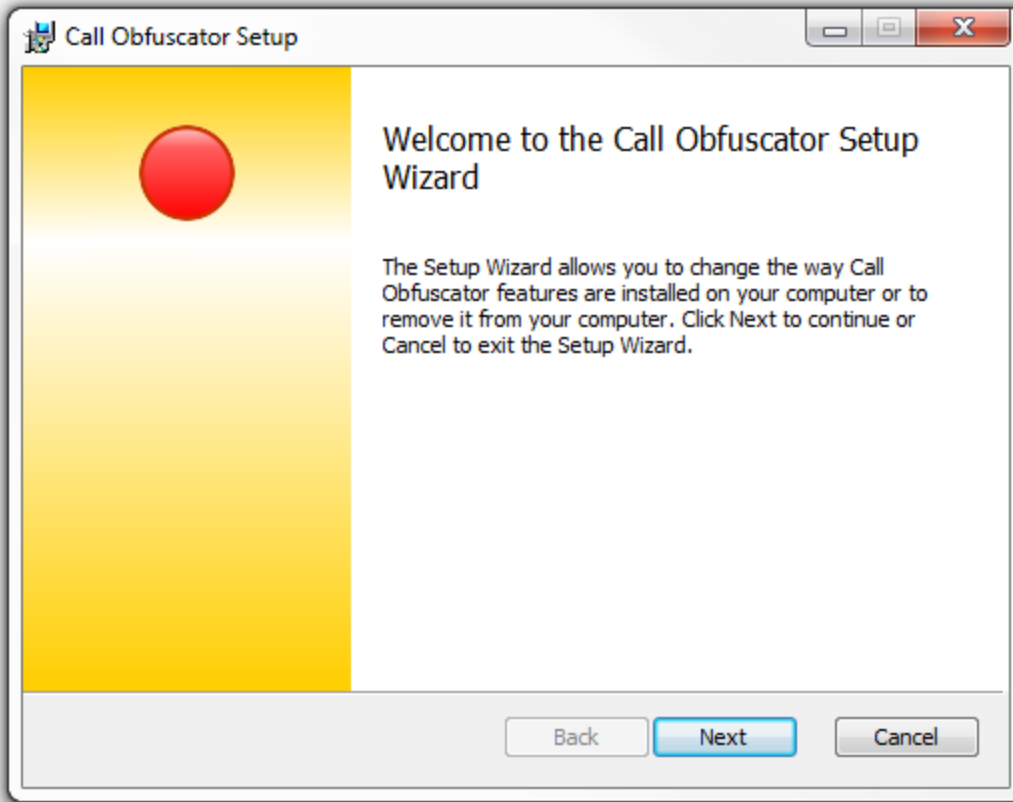
> ℹ️ A <default> element can be added as the fall-back tone to use.

Call obfuscator

> ℹ️ 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.

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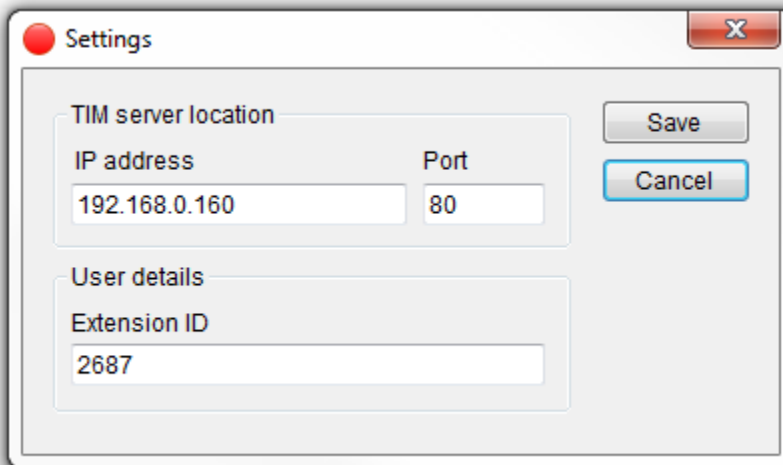
Field	Description
IP address	The IP address of the PC running TIM Plus

Port	The port number that TIM Plus is listening on
Extension ID	The extension number of the user whose calls should be obfuscated by this instance of the Call Obfuscator

A web user with the following credentials must exist in the call logger, in order to authenticate the `http` request sent by the application:

username: `admin`
password: `tntnet`

1. During a phone call, start the obfuscation by left-clicking on the icon which, after some milliseconds, will change to , 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.