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1. Introduction

1.1. Product Overview

CallRecorder is an easy to use VOIP call recording solution that implements the corporate call recording keeping policy and provides secure and easy access to call records.

It allows managers to review and score phone calls according to their work group. Users are empowered by providing them with accurate records of their calls.

CallRecorder is a self-contained software recorder which includes everything necessary to record VoIP calls (besides the Operating System and server hardware): database, web server, Java, etc.

1.2. Features and Benefits

- **Automatically Record Phone Calls** - Damage control and increased accountability in your personnel, suppliers and customers.
- **Manual (On-Demand) Control**: recording can also be initiated by the user using the browser or IP Phone Service. Both Full Call and Partial Call recording modes are supported.
- **Multiple Recording Methods**: CallRecorder supports both port mirroring (SPAN) and forked recording (SPANless). Hardware recording coming soon!
- **Monitor live calls**: Listen to a call in progress, through the browser or IP phone service. You can also whisper to the agent without being heard by the external party (IP phone service).
- **Browse Recordings by Agent** - Superb browsing interface tracks agents across multiple phone numbers.
- **Search** by caller ids, phone numbers, annotations, time, description, tags, etc.
- **Replay, Annotate and e-Mail** phone call recordings. Easy, secure access to call recordings, using a web audio player, desktop audio player or IP Phone Service.
- **Tagging & Searching** calls with customized hierarchical tags has never been easier.
- **Call Scoring & Custom Forms** - Integrated agent scoring and reporting module
- **Agent Call Statistics Reports**: the number of calls made, received, etc by each agent
- **Email Notifications**: automatically send email when a predefined number appears in a call.
- **Backup & Restore**: Archive calls on DVDs, HD-DVDs, BluRays or SANs. A single DVD can store up to 15,000 calls of 5 minutes each, due to the state-of-the-art voice compression technology incorporated in CallRecorder.
- **Reverse Caller Lookup** - Displays the caller name and business unit using the company Phone Directory.
- **Multi Site Replication**: using queued replication, you can record many network partitions and centralize recordings at the HQ. Recording and replication survive a downed WAN link.
- **Screen Recording** – Integration with Memolith Screen Recorder. See what was done on the screen while the phone call occurred.
- **Specialized Speech Compression** lowers the storage requirements 8 times over MP3 and allows 18,000 hours of phone calls storage on one 120 GB hard drive.
- **Call History** - follow a call as it is transferred, put on hold or parked
- **Audit Replays** – prevent recordings abuse by browsing the list of accesses to a call.
- **XML Phone Service** – handily review your past calls from your XML enabled phone (Cisco IP Phones 7940, 7960 & 7970). Authenticate, Browse, Play, Rewind, e-mail, mark important. You can also assign calls to folders.
- **Access Control Lists** – fine grained permission system to allow listening and acting on calls.
- **Automatic Software Update** - the easiest, fastest way to apply patches
- **Thin Client Deployment** – The administration and user tools run in all web browsers supporting Flash.
- **Integrated Support Tools** – Request & receive technical support with a few clicks, by using the integrated log packer and TeamViewer support tool.
- **Passive network sniffing** assures zero impact of recording on PBX performance and improves system reliability.
- **Try Before You Buy** – Download a fully-featured evaluation version with a friendly configuration wizard from: http://www.call-replay.com

### 1.3. Technical Specifications

| **VoIP PBX** | • Cisco CallManager (all versions)  
|              | • CallManager Express  
|              | • Avaya CM S8000 series and IP Office 500  
|              | • NEC Univerge - SV8000 series, IP only  
|              | • generic SIP  
|              | • IPTrade turrets  
|              | • Mitel  
| **IP Phones** | • All Cisco IP Phones  
|              | • All SIP phones  
| **Operating System** | • Any Windows OS  
|              | • 32 and 64 bit compatible  
| **Hardware Requirements** | • Software only recording system, no proprietary cards  
|              | • Industry-standard Intel compatible server supplied by customer  
|              | • Network connection to voice traffic, using a hub or a mirrored port for promiscuous mode network sniffing  
| **Recording Capacity** | • Up to 400 simultaneous calls on a single dual core CPU  
| **Retention Capacity** | • Speech compression, VBR, Stereo, 170 hours per GB  
|              | • ~ 18,000 compressed talk hours on one 120GB HDD  
| **Supported codecs** | • G.711, G.722  
|              | • G.729 (extra option)  
| **Recording Architecture** | • Passive network sniffer, Skinny Protocol  
|              | • SIP trunk recorder compatible with newer Cisco phones  
|              | • Stereo, each party is heard in a different channel  
| **Embedded Database** | PostgreSQL 9.0  
| **Security** | • Secure access to recordings  
|              | • Managers have access to calls based on logical departments filters  

| Call records access                                      | • Web interface + desktop player  
|                                                        | • Phone Service interface (on Cisco IP Phones 7940, 7960, 7970) |
| Support                                                | • Technical Support includes Software Upgrades  
|                                                        | • TeamViewer software included |
2. Prerequisites

2.1. Hardware Requirements

Network

SPAN recording requires:

- Managed network switch with port-mirroring capability (SPAN)
- At least two network interface cards are required on the recording server, one for each monitored switch, and another for the site (a monitoring port can only receive packets). Using one interface card is possible, but the administration site will only be accessible from the physical console.

Forked recording requires:

- One network interface card on the recording server, for administration site and recording
- Cisco CallManager platforms newer than 5.0. For Cisco CallManager Express please use SPAN recording.
- Selected Cisco phone models.

Server

Virtual Machines: VMs are supported in all recording modes for up to 50 simultaneous calls. Forked Recording mode works directly, while SPAN recording requires additional configuration to enable network cards to work in promiscuous mode.

CPU: any modern quad core CPU will support 500 simultaneous calls, the limits are in the RAM and disk IO subsystems.

RAM: 1 GB of RAM for the system + 1GB for each million call records stored in the database. For example, if you intend to store 2 million calls in the db, provision at least 3GB of RAM for the server.

Storage: audio compression rate of 1.7 KB/s means that each GB on the drive can keep about 160 hours of recordings (voice compression is 8 times better than MP3). That is, you can keep 96,000 calls of 10 minutes on one 100 GB HDD. One month of typical call center recordings (160,000 calls) takes about 55 GB.

Please use only redundant disks (RAID1 and above) for storing call recordings!

2.2. Software Requirements

Server Side

Operating System: Any Windows Server. Desktop Windows OSes (Professional, XP, Vista, 7) can also be used, but TCP connection on these platforms are limited to 10 simultaneous sessions, limiting the number of users which can replay calls using the web site or a phone service.

IMPORTANT: It is recommended not to run any other server software on a production recording server! Call recording is essentially real-time. Failing to keep up with the traffic can result in lost calls. Other software can unpredictably use critical processor and memory resources which are necessary for packet capture. The application allocates computing resources according to priorities so an uncontrolled processing spike of another program may disrupt recording.

Client Side

- Any operating system, with a browser running Adobe Flash 11.1 or later, (for example Internet Explorer 6,
Software Optimizations

The Windows system cache is by default too large. Go to Control Pane / System / Advanced / Performance Settings / Advanced:

- set memory usage for best performance of Programs
- set processor scheduling for best performance of background services

2.3. Get the application and the license

Application

Our website always contains the latest version of application:  http://call-replay.com/go/download
The setup automatically downloads the latest patch from our web site. If that does not happen MAKE SURE you download and apply it manually.

License

If you want an evaluation license, you can request it via email from info@call-replay.com or http://call-replay.com/go/contact.
3. Installation
Setup on Windows

3.1. Welcome Screen

Make sure you have read the Prerequisites chapter before proceeding further.

1. Check if the setup version you are running is the latest available from our web site.
2. Click Next

---

3.2. License Agreement

1. Click "I Accept" if you agree with our license. You may not use the software if you do not agree.
2. Click Next
3.3. Destination Folder

1. Select the destination folder for the application binaries (calls are stored separately);
2. Click Next.
3.4. Download the latest patch

It is very important to have the program up-to-date so please leave the option marked unless the firewall blocks connections to the internet. The web site used for updates is update.call-replay.com. If your computer does not have access to the Internet, deselect "Download update" and apply the patch manually after the setup has completed. To get the latest patch, go to http://www.call-replay.com/go/dl/.

![Download Latest Update dialog box]

3.5. Data folder for storing call recordings and database files

This is the folder where all the user data, including audio files, the database, logs and licenses will be stored. Please select the disk with the maximum available free space. Use a RAID protected disk for this folder.
Site configuration

At the end of the setup, the Site configuration utility will be automatically started to assist configuring of the name, IP and port of the administration web site. You may use port 80 if it is not used by another web server.

**The IP of the website is also used for licensing purposes.**

Press OK to start the administration web site.

3.6. Database Setup

This page appears only if the database does not already exists in the data folder. You have two options:

a. If this is a fresh installation (not an upgrade), the left-side panel should be used. Enter the administration account and password for the new database, then click the Create Database button.
b. When upgrading an existing installation of version 5, enter the location of MSDE database server, the name of instance, database name, user name and password, then click Upgrade Database.

3.7. Login page

Enter the user name and password. When running with an evaluation license, this page always displays the name of an administrator and a public password "eval" which works for every user.
Setup on Linux

At the moment, the only supported platform is:

- Ubuntu 12.04 LTS Precise Pangolin x64

Online Installation

1. Download and install the repository public key
   
   ```bash
   wget -q http://www.call-replay.com/apt/RAISoftware.asc -O- | sudo apt-key add -
   ```

2. Add the appropriate repository line in your `/etc/apt/sources.list` file
   
   ```
   deb http://www.call-replay.com/apt wheezy non-free
   deb http://www.call-replay.com/apt precise non-free
   ```

3. Update and install CallReplay
   
   ```bash
   sudo apt-get update && sudo apt-get install callreplay
   ```

Offline Installation

1. Download the Ubuntu Linux x64 version from the CallReplay site
2. Install dependencies
   
   ```bash
   sudo apt-get install openjdk-7-jre postgresql postgresql-contrib libpcap0.8
   p7zip-full speex sox libav-tools dmidecode
   ```

   Optional packets for desktop environments:
   
   ```bash
   sudo apt-get install flashplugin-installer pgadmin3
   ```

3. Install CallReplay
   
   ```bash
   sudo dpkg -i callreplay-*.deb
   ```
Migration to a New Server

When you want to move your existing installation to a new server but preserve all data, please follow the steps:

**Easiest Procedure**

1. On the old server, shut down the CallReplay and CallReplayDb services and set their Startup Mode to Disabled.
2. Copy or move the CallReplay Data Folder to the new server. The CallReplay Data Folder is configured once at installation time and contains the sub-folders Database, Calls, Licenses, Logs, etc.
3. Then run CallReplay Setup on the new server. When asked about the Data Folder to use, point it to the copied folder containing old data.
4. After setup completes, you should be able to see the old calls and play them.
5. Cleanup the old server. Do not start the old server again after the license was re-activated on new server.

**Minimum Downtime Procedure**

1. Install and configure CallReplay on the new server. Verify that recording works.
2. Copy or move the CallReplay Calls folder to the new server, into a new folder. Exclude from copying/moving the sub-folder of current day.
3. On the old server, shut down the CallReplay and CallReplayDb services and set their Startup Mode to Disabled.
4. Copy/Move the Database folder to the new server, then copy/move the folder left un-copied at step 2.
5. On the new server, shut down the CallReplay and CallReplayDb services.
6. In the CallReplay Data Folder, move the Database and Calls folders to a backup folder.
7. Move in their place the Database and Calls folders copied at steps 2 and 4.
8. Start the CallReplay and CallReplayDb services.
9. Review recording configuration (it has the config from the old server) and verify functionality.
10. If necessary, Restore the Calls folder backed-up at step 6.
11. Cleanup the old server. Do not start the old server again after the license was re-activated on new server.
4. Managing Licenses

Before the call recorder can be used you need to add a valid software license file.
If you need an evaluation license, please send an email at info@call-replay.com or check our contact page http://call-replay.com/go/contact.

There are two kinds of license files:

- .lix is a generic license, not-activated or bound to any computer.
- .bind is an activated license file which only works on the computer where it was activated. Only production licenses need to be activated.

A license file may only be used on one computer at any given time. Loading the same license file (that is, with the same license ID) on two servers at the same time is a violation of the usage agreement. Also a violation is using the license file after it has been revoked or superseded by a newer license file.

To add a new software license file, select System / Licenses from the menu then click the Upload button. The license fill will be automatically activated if necessary and stored in the DataFolder/Licenses directory.

If the licensing software cannot contact our web site for license verification, please open the Manual Activation link (http://lix.call-replay.com/Lix/ManualActivation/) on a computer with Internet access and fill in the requested information.

4.1. Company Wide Recording (CWR) License

This licensing model is obsolete, do not configure it unless instructed by technical support. If you have a Company Wide license type, ensure that application can contact CallManager to establish the number of registered phones.

You must supply a user name and a password in the “Company Wide Recording” tab, which is used for authentication with the CallManagers. This user must be the same on all the specified CallManagers. The password must be the same on all the specified CallManagers. You don't need to specify the password each time that you make an update in the configuration page, but only when you need to change the existing password.
If the number of phones registered in your CallManager exceeds the number of registered phones in your licenses, the application service will not be started.
5. Configuring Recording

Overview

CallReplay Call Recorder offers two methods for recording calls: Forked (SPANless) and SPAN recording. The modern, recommended, recording method is Forked Recording, because of its ease of installation.

For Cisco CallManager Express please use SPAN recording, and manually set the PBX type to Express, as auto-detection will not work.
5.1 CallManagers (PBXs) Configuration

PBXs Configuration Page allows management of Cisco CallManagers, Mitel, IPTrade, NEC and other supported telephony controllers.
You can add a PBX by clicking on the Capture / CallManagers (PBXs) / Add PBX.

1. Add all the IPs of CallManagers in a cluster
2. Set its type to either Cisco CallManager, IPTrade, Mitel or SIP.
3. Enter the PBX version.
4. When using Cisco UCM > 5.0 you have the option to choose between passive and active (forked) call recording.

Observation: Cisco CallManager Express is a different type than Cisco CallManager.
5.2 Forked Recording

Forked Recording

Forked Recording (SPANless Recording) is an active recording technology, available only on Cisco CallManager platforms newer than 5.0 and selected Cisco phone models.

Benefits of Forked Recording

Ease of use and management

- Establish complex network architectures not depending on SPAN ports
- Move or reconfigure complete departments with ease
- Improve control over branch locations

Economical

- Reduce OPEX with easier administration as no configuration of SPAN ports is necessary
- Reduced CAPEX – need for fewer elements at the branches

Reliable and secure

- Free-up resources for network monitoring
- Increase reliability utilizing system resources better and more manageable
- Security: Both authenticated and encrypted mode can never be recorded.
- Geo redundancy available with high bandwidth utilization

Additionally

- Internal calls recording = RSPAN, VLAN split to meet SPAN capacity
- Cisco plans support for CUCM 8.x
- Built-in support of recording notification tones

How does it work?

The Cisco Unified Communications Manager (CUCM) interface provides two recording modes:

- Automatic recording recording all calls on line appearance. This method is invoked by CUCM.
- Selective recording allowing users to record ad-hoc or also allows recording server to record based on business rules and events.

After calling-in and routing the call to an agent CUCM automatically sends two call setup messages to the Agent device. The 1st call is the agent stream and the 2nd call is customer stream. The Communications Manager invites the recorder to both calls via SIP Trunk and the recorder accepts both calls and receives RTP streams from Agent device.

Forked recording requires:

- usage of the silent monitoring and recording interface of CUCM
- CUCM version 6.0 and higher
- the usage of 3rd generation phones, as detailed here: http://developer.cisco.com/web/sip/wikidocs/-/wiki/Main/Unified+CM+Silent+Monitoring+Recording+Suppo}
Usage notes

Usage of a SPANless configuration will bring major benefits as long as the following requirements and notes are taken into account:

- The expected increase of network traffic
- For PSTN recording only GW span works as simple trunk recording
- TAP switches are able to handle high traffic in large architectures with centralized GW
- Only 3rd generation phones are supported
- There is no support for active-active redundancy
- Interruptions in recording may occur if a failure occurs during the call
- If the WAN capacity is limited, redundant recording may be refused (due to automated network intelligence)


Configuration Steps

In the Capture / PBXs page, the Forked Recording wizard button appears only when the PBX type is Cisco CallManager, version is greater or equal than 5.0 and the Forked Recording Protocol is selected in the PBX options. In this case other recording protocols should be disabled.

Click the button labeled Forked Recording to start the configuration wizard.

1. The following configuration wizard utility will be shown. Fill in the CUCM administrator user name and password. Press **Connect** button.
2. In the next page leave all the fields unchanged unless instructed by technical support. The Phone Service IP address must be routable from the phones’ VLAN. Press **Continue** button.
3. Here you must select the correct CallManager Group, Recorder Extension for CallReplay, the IP address of the CallReplay recorder (also needs to be routable from a phone's IP address), CSS and Device Pool for recorded phones, Security Profile, SIP port of Callreplay (Syn-Apps' SA-Announce also uses a SIP trunk so make sure in that case to select another port, such as 5061) and the RTP port range used by CallReplay for receiving audio streams. The recorder extension must have the same number of digits as recorded extensions. Press Next button.
4. Here you have a list of a non-managed phones (left side) and the list of managed phones (right side). Select from the non-managed list the devices you want to be recorded and add them in the the managed phone list. When a managed phone has the Record check box selected it will be recorded. Otherwise the device will NOT be recorded even it is in managed phone list. Similarly for Service check box. When that check-box is selected, the phone will be subscribed to CallReplay Phone Service (see step 2), which will become available on the Service button (on the phone). Thus we can control recording and monitoring independently for each phone. You can filter the phones by any of the columns displayed, i.e. Phone Name, Description or Extension.
Press **Continue** button and the selection will be processed. The status of processing will be shown.
Press Close button to close the wizard configuration utility.

**Cisco CallManager - alternative configuration for Forked Recording**

We recommend the usage of CallReplay Forked wizard for configuring your Cisco CallManager. But if you do not want to use this feature follow these steps to prepare your Cisco CallManager for recording.

1. **Recording phones using forked**

You need to create by hand an application user - *CallReplay*, a new SIP profile - *CallReplay SIP profile*, a recording profile - *CallReplay recording profile* and a new trunk - *CallReplay_SIP_trunk*.

**CallReplay Application User**

In CallManager site administration go to User Management > Application User. Press *Add New* button and fill in the text fields as follow:

- **Application User Information**
  - **User ID**: *CallReplay*
  - **Password**: *password*
  - **Confirm Password**: *password*
  - **Presence Group**: select *Standard Presence group*
**Device Information**
Add from the list of Available Devices to the list of Controlled Devices the phones who will be recorded. If you use EM profiles then add from "Available Profiles" list to the "CTI Controlled Device Profiles" list the EM profiles who will be recorded.

**Permissions Information**
Groups: press Add to User Group button and in list shown subscribe to the following groups: Standard CTI Enabled, Standard CTI Allow CallRecording, Standard CTI Allow Control of Phones supporting Connected Xfer and conf and Standard CTI Allow Control of Phones supporting Rollover Mode

Press Save button to create the application user.

**CallReplay SIP profile**

In CallManager site administration go to Device > Device Settings > SIP Profile. Press Add New button and fill in the Name* with value CallReplay SIP profile. Save the changes

**CallReplay recording profile**

In CallManager site administration go to Device > Device Settings > Recording Profile. Press Add New button and fill in the text fields as follow:

Name*: CallReplay recording profile

Recording Calling Search Space: select a CSS

Recording Destination Address*: xxx - the extension used by CallRecorder. The value must be an unasigned EXT and having the same number of digits as recorded extensions. If extensions in your network have 3 digits then this value must have 3 digits too.

Save the changes.

**CallReplay SIP trunk**

In CallManager site administration go to Device > Trunk and press Add New button to define a new trunk. Fill in the text fields:

Device Information

Device Name*: CallReplay_SIP_trunk_xx.xx.xx.xx where xx.xx.xx.xx is the IP of CallRecorder server.

Device Pool*: select the device pool where the recorded phones are registered

SIP information

Destination Address: the IP of the CallRecorder server

Destination Port: the port where CallRecorder is listening (default value 5061)

SIP Trunk Security Profile*: select security profile accordingly

SIP Profile*: select CallReplay SIP profile (created before)

Save the changes.

**2. Subscribe phones to the CallReplay Call Recorder Phone Service**

Create CallReplay Call Recorder phone service

In CallManager site administration go to Device > Device Settings > Phone Services and press Add New button to define a new phone service. Fill in the text fields as follow:

Service Name*: CallReplay Call Recorder

ASCII Service Name*: CallReplay Call Recorder

Service Description: CallReplay Call Recorder
Service URL: http://IP:PORT/CallRecorder/phoneService (you can get these values from CallReplay Site Configuration);
- Service Category*: select XML Service
- Service Type*: select Standard IP Phone Service
- Enable: True

Save the changes.

Subscribe devices

In CallManager site administration go to Device > Phone and perform the following operations:
- select a device;
- select from Related Links dropdown list option Subscribe/Unsubscribe Service and press Go button;
- in the new windows select CallReplay Call Recorder service
- press Next and after that Subscribe button;
- close the window;

After the phone reset the CallReplay Phone Service becomes available for use.
5.2.1 Encrypted Recording

1. Pre-requisites

2. Install

3. Configure certificates

1. Pre-requisites

To be able to record encrypted calls there are the following pre-requisites:

- Call Manager set to Mixed Mode

- Each phone that is needed to have encrypted conversations to be set up with a Secure Profile

2. Install

The steps required for encrypted recording are almost identical with the steps required for normal forked recording. Follow steps 1-2 from normal forked recording (Forked Recording).

At step 3, to be able to record encrypted calls, we need a secure SIP trunk.

If this is the first time that you are configuring Encrypted Recording, you must choose "New Secure Trunk Profile" from the Security Profile list,

otherwise choose "CallReplay Secure Trunk Profile":


We recommend that you choose 5061 as the SIP port as it is the default value for secure connections.

From this point on you can follow the normal forked recording manual (Forked Recording).

Note: You can choose the Encrypted Recording Profile for both encrypted and unencrypted calls. The same profile and trunk can serve both protocols.

3. Certificate configuration

For security reasons, Call Manager sends data that involves secure calls only on a Secure Trunk.

Both Call Manager and CallReplay need to exchange certificate information in order to establish a secure connection.
1. Importing Call Manager's Certificate on CallReplay

Log on to **Cisco Unified Operating System Administration**. From **Security -> Certificate Management** download the Certificate with **Certificate Name**: **CallManager** and **Certificate Type**: **certs**.

Download and save the certificate to a known location. Import this certificate through the CallReplay's Web Wizard (System->Certificates->Import Cisco Certificate)

![Image](https://example.com/image.png)

**Cisco Certificates**

- Change Certificate...
- Export Certificate
- Import Cisco Certificate

On the next dialogue choose the file downloaded earlier.

2. Exporting CallReplay's Certificate to Call Manager

1. Generate the certificate
   
   If this is the first time that you configure CallReplay's certificate, you need to generate a new certificate.

   From the CallReplay's Web Wizard (System->Certificates) press the button "Change Certificate". On the next dialogue choose "Generate self-signed certificate".

   If the generation was successful, a new information will appear, containing various information about the new certificate.

2. Export certificate

   Export the newly created certificate by clicking "Export Certificate" button. Save the certificate to a known location. This certificate will later need to be imported in Call Manager.

3. Import the certificate into Call Manager

   Log on to **Cisco Unified Operating System Administration**. From **Security -> Certificate Management** choose "Upload Certificate/Certificate Chain".

   In the next dialogue choose as **Certificate Name**: **callmanager-trust** and upload the file saved in Step 2.
4. After all certificate operations are completed successfully *reset* "CallReplay SIP Trunk" (in CallManager go to: Device -> Trunk -> CallReplay SIP Trunk * and choose *Reset not Restart*)
5.3 SPAN Recording

Overview

SPAN Recording is a passive technology. It is working with all Cisco CallManagers and all kind of phones as long as they are SIP or SKINNY compatible.

The application service uses a network interface card functioning in promiscuous mode, in order to capture packets for the conversation recording. The host computer or the server need a network connection to voice traffic, through a non-switched hub or through a SPAN port on a switch. For more information about configuring a mirrored port on your switch, read the user manual of the switch.

You can also visit the following links:


The application works by monitoring phone traffic. There are two types of phone traffic essential to recording:

- signaling (call control), from phones or voice gateway to CallManager
- audio streams (RTP), from phone to phone, or from phone to voice gateway

To be able to record calls, the application needs to intercept both types of traffic, call control and audio streams. For more information about network sniffing please read the Wireshark FAQ: http://www.wireshark.org/faq.html

Virtual Environment

The difficulty in getting SPAN based/passive recording functional in a virtual environment is the SPAN itself. With the SPAN configured on a physical switch port, all communication will be directed to a physical NIC. This physical NIC MUST be bound on the VM. Thereason behind this statement is because the most virtual NIC software cannot forward the SPAN information to the VM, so the physical NIC will be required to be bound to the VM. Because of this requirement installing application into a virtual environment may not be advisable. Allocating a physical device to a VM only requires VT-D support in the host CPU.

Implementation Options

There are two main options in configuring network monitoring:

1. **Record only external calls.** This is the easiest. All you need to do is to have the CallManager and the voice gateway in the same switch and SPAN them to application recording port. If you have them in different switches you need two monitoring NICs in the recording application server, one for each switch. **Make sure you are mirroring all the CallManagers and voice gateways, including backup ones.**

2. **Record all calls, external and internal.** For that you will need to have all the phones' traffic monitored to the application recording port. That is, monitor all switches with phones. Usually this is done by placing all the phones in a separate VLAN, and monitoring that VLAN. Using a VLAN also has the advantage of discarding the general (PC) network traffic, which can overload the monitoring interface in both the switch and in recording application.
Because RTP traffic (green and red in the figure) travels directly between end points, without going through CallManager, in a large enterprise with multiple switches only external calls (PSTN) can be recorded in a cost effective way (one recorder for each voice gateway).

To record internal calls also, one recorder per switch with IP phones is required.

**Server Side Network configuration**

Because monitoring ports cannot usually transmit traffic, for production setups the server you are using must have a minimum of two network cards for application to function properly. One of them will be used for general network traffic and accessing Application Web Administration Interface and the other for listening VOIP related traffic. The NIC selected for website access should have a static IP address.

**Important:** The monitoring NIC should not have a routable IP address, because network switches inhibit the transmission of packets from ports used as port mirroring destinations. If the OS routes packets through this port, all sent packets will be discarded. Remove the IP address of the monitoring NIC, or set it a non-routable address, such as 1.1.1.1.

We assume that you have already configured your network switches in order to mirror all VOIP traffic from VOIP LAN to the monitoring NIC of CallReplay otherwise no calls will be recorded.

Open the **Capture / Network Interfaces** page, and choose which NIC is used for VOIP traffic and which one is used for web administration and general traffic.
If you have configured port mirroring correctly, calls will start being recorded immediately. For CallManager Express please see the PBX Configuration chapter.
5.4 IPTrade Turret Recording

Active Recording Settings

This section represents the system configuration for active recording on IPTrade turrets. Administrators must set the CallReplay as the recording server and indicate that server to the IPTrade system. The following settings have been confirmed to work on CallReplay v7.5 and IPTrade Turret Support Server (TSS) v7.4.24448.

Call Replay settings

On Recording > PBX page choose New PBX and from the PBX Type, select IPTrade Active.

Fill in the the IP Addresses field with any address (ex. 0.0.0.0).

From Protocols list choose IPTrade Active and edit the protocol settings:

- Port indicates the TCP port to listen for incoming connections. Default value is 4456. You must enable this port into the system firewall.
- VAD indicates whether or not the recording server will use Voice Activity Detection settings for silence detection.

Turret Support Server settings

Following settings are required on TSS for integration with the active recording system.

1. Set turret recording engine
   Go to System\Settings\FTP settings (turrets) and on Recorder - Basic Mode choose "iptrade" for Recorder compatibility.

2. Configure the voice recorder address
   Use the same location as the one used to set the recording engine or inside user profile (Account Management\User) or shared profile (Account Management\Shared Profile) go to Settings tab and on Recorder - Basic Mode configure the voice recorder location in form <protocol>://<address>:<port> where :

- Protocol indicates the voice recording system to be considered. Valid values are : vrc.
- Address indicates the IP address or the name of the voice recording service host.
- Port indicates the TCP port to connect to the voice recording service (same as the value used to configure the IPTrade Active Recording protocol in CallReplay).

Example : vrc://192.168.0.1:4456
5.5 Mitel SRC Recording

1. Overview:

The Secure Recording Connector (SRC) service is a software solution that facilitates the recording of Mitel encrypted voice streams by third-party call recording equipment (CRE). The Mitel Border Gateway (MBG) server with SRC service is positioned on the LAN between the ICP and the sets to be recorded. It accepts requests from an authorized CRE to establish taps in the voice stream. These taps are separate (mirrored) streams from the SRC to the CRE.

CallReplay offers a native solution for recording calls on Mitel PBXs for both direct and indirect call recording setups.
There are several abbreviations that will be used throughout this manual:

SRC - Secure Recording Connector
MBG - Mitel Border Gateway
MCD - Mitel Communications Director
ICP - Integrated Communications Platform

2. Requirements:

1. Mitel SRC Licenses

SRC Licenses are a Mitel requirement and need to be acquired from your Mitel Partner. You need as many licenses as CallReplay Licenses.

When the licenses are installed, you should be able to see the number of Tap Licenses in **MBG -> Status -> Dashboard**.

3. SRC configuration and phones setup

![Manage Mitel Border Gateway](image-url)
1. For CallReplay to be able to receive audio streams, each phone that needs recording, has to be routed through the MBG (Mitel Border Gateway). This can be achieved in two ways:
   a. Register the phone to the MBG
   b. Set phone's Gateway to be the address of the MBG

2. In each MCD or ICP 3300 you need to allow HCI Call Control and Monitor.

   This can be achieved in the section System Properties -> System Feature Settings -> Class of Service Options.

   Program this setting for the COSs (Class of Services) of the phone sets that need recording.

4. CallReplay Setup

1. CallReplay Wizard

   For CallReplay to be able to communicate with your PBX, it needs to know some information about your network set-up.

   - In PBXs menu, there are the following options that you need to set:
   - IP Addresses: The IP addresses of your SRC (usually the same IP as the MBG).
   - PBX Type: This should be set as Mitel MCD 3300
   - PBX Version: The version of your MCD or ICP (eg. 6.0)
2. Accept CallReplay Certificate in MBG Certificate Management:

![Certificate Management Interface]

- **Operation status report**: Certificate ID 35126688-ef5-5-566-e-127a7c7082 was successfully received.
- **Issued**:
  - Issue: CA="CA, ST=DK, O=Mbit Networks, OU=VolP, CN=Mbit 600 C (Mbit600)
  - Subject: CN=ServicesLib Account ID: 2264277644/emailAddress=admin@mgb.com
  - Not Before: Apr 9 15:13:00 2014 GMT
  - Not After: Apr 9 15:13:00 2014 GMT

**Quashed CSRs**
- **Certificate ID**: e79a9b3d-cb14-4901-8494-5e5e92d973c
- **Subject**: CN=CallReplay

**Approved Certificates**
- **Certificate ID**: e79a9b3d-cb14-4901-8494-5e5e92d973c
- **Subject**: CN=Mbit mgb.com

---

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CallReplay certificate should be in the Queued CSRs.

Note: It might be required that you restart the CallReplay Service

3. Set the phone sets that you want to record.

Switch back to CallReplay Wizard interface. In the PBXs menu, on the entry created in Step 1, click **Active Recording**, and choose the phones that you want to be recorded as shown in the following image:
6. Testing the Installation

Make sure there are no pending alerts below the menu bar, by clicking on each alert and solving them.

6.1. Playing a Call

Make a test call

Place a call to an external number. Recording internal calls requires a special network configuration, please see the Network Configuration chapter.

Playing a call

1. From the menu, choose Recordings / Replay Calls as shown below:

2. If there are no calls displayed please skip to the “Troubleshooting” chapter.
3. Click the More button (green plus sign on the right side of every call), choose Properties and a new window, named "Call Details" will appear, as shown below.
   This page contains all information about a call like caller party, called party, duration of call, file size, file format. In the Advanced tab, you will get information about RTP traffic (ip address and port used in recording call).
   In the Description tab, you can provide a description for that call. In the Export tab by pressing "Save Call" you can save the call under Wav format or speex format. You can email that call by pressing "Send Email" button.
1. Click on Play Button.
2. A pop-up will be open and the recording will start playing.
Attention You must have a valid G.729 license to play a G.729 recording, otherwise an error will be displayed.
7. Troubleshooting

7.1. Analyze Packets captured

By clicking **Capture / Troubleshooting**, you can see the amount of TCP or UDP Packets captured from and to an IP Address.

![Capture Troubleshooting](image)

### Table: Packets Captured

<table>
<thead>
<tr>
<th>IP Address</th>
<th>MAC Address</th>
<th>TCP</th>
<th>UDP</th>
</tr>
</thead>
<tbody>
<tr>
<td>10.1.1.105</td>
<td>00:17:41:14:1f:45</td>
<td>30</td>
<td>0</td>
</tr>
<tr>
<td>77.77.177.232</td>
<td>10:11:00:30:30:30</td>
<td>0</td>
<td>48</td>
</tr>
<tr>
<td>10.1.1.123</td>
<td>62:54:00:60:06:56</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>193.1.292.124.277</td>
<td>10:11:00:00:10:01</td>
<td>0</td>
<td>24</td>
</tr>
<tr>
<td>10.1.1.253</td>
<td>80:11:00:01:43:03</td>
<td>0</td>
<td>2</td>
</tr>
<tr>
<td>131.234.137.24</td>
<td>10:11:00:00:10:01</td>
<td>0</td>
<td>24</td>
</tr>
<tr>
<td>10.1.1.255</td>
<td>00:00:00:00:00:00</td>
<td>0</td>
<td>2</td>
</tr>
<tr>
<td>10.1.1.188</td>
<td>00:FF:76:36:51</td>
<td>0</td>
<td>126</td>
</tr>
<tr>
<td>911.205.8.36</td>
<td>10:11:00:00:10:01</td>
<td>0</td>
<td>24</td>
</tr>
<tr>
<td>10.1.1.52</td>
<td>00:17:41:14:1f:45</td>
<td>236</td>
<td>0</td>
</tr>
<tr>
<td>224.0.0.250</td>
<td>01:00:00:00:00:00</td>
<td>0</td>
<td>1</td>
</tr>
<tr>
<td>10.1.1.101</td>
<td>00:17:41:14:1f:45</td>
<td>104</td>
<td>0</td>
</tr>
<tr>
<td>224.0.1.178</td>
<td>01:00:00:00:00:00</td>
<td>0</td>
<td>1</td>
</tr>
<tr>
<td>10.1.1.156</td>
<td>00:17:41:14:1f:45</td>
<td>7563</td>
<td>0</td>
</tr>
<tr>
<td>10.1.1.158</td>
<td>00:17:41:14:1f:45</td>
<td>0</td>
<td>48</td>
</tr>
<tr>
<td>193.2.292.271</td>
<td>10:11:00:00:10:01</td>
<td>0</td>
<td>24</td>
</tr>
<tr>
<td>10.1.1.91</td>
<td>00:17:41:14:1f:45</td>
<td>6264</td>
<td>294</td>
</tr>
<tr>
<td>239.2.11.71</td>
<td>01:00:00:00:00:00</td>
<td>0</td>
<td>126</td>
</tr>
<tr>
<td>5.5.5.53</td>
<td>10:11:00:00:10:01</td>
<td>0</td>
<td>24</td>
</tr>
<tr>
<td>46.29.177.23</td>
<td>10:11:00:00:10:01</td>
<td>0</td>
<td>48</td>
</tr>
</tbody>
</table>

7.2. Analyze the Network Configuration

**Install Wireshark**

Wireshark is a free, open source packet analyzer.

1. Download the latest version from [http://wireshark.org/download](http://wireshark.org/download)
2. At installation, be sure to **not** select the "Install WinPcap" option (as it is already installed by CallRecorder) or the "Start WinPcap service "NPF" at startup" option.
Capture a Call

1. Go to the **Capture** menu and select **Interfaces**
2. In the interfaces window select the **same** network card you configured for monitoring in CallRecorder.
3. Press the **Capture** button

4. Start the call
5. Talk a few seconds. In the capture window you should see the number of udp and tcp packets growing. If not, you either configured SPAN incorrectly or captured on the wrong NIC.
6. Stop the call.
7. Stop the network capture.

Usually there should be about 100 udp packets per second, so if this number is much smaller, then your server may not be capturing the voice stream.

Verify Call Control

In the main **Wireshark** window, set the filter to:

```
ip.addr == {TEST_PHONE_IP} && skinny.messageid == 0x8a
```
and press *Apply*.

The `StartMediaTransmission` and `OpenReceiveChannelAck` messages are critical for recording to work.

If you don't see any skinny packets after applying this filter, then your call control stream was not captured. To fix it, you should configure SPAN to your computer, in order to forward traffic from **CallManager**.

### Verify the Voice Streams

In the main window, set the filter to:

```plaintext
ip.addr == {TEST_PHONE_IP} && udp
```

and press *Apply*

If you don't see any RTP packets after applying this filter, then your voice stream was not captured. To fix it, you
should configure span to forward traffic from your phones to your computer.

If you are only interested in external calls, you should only forward traffic from your voice gateway to your computer.

7.3. GET Remote Technical Support

The recording application has a number of powerful support utilities included. Team Viewer for eg., is a screen sharing utility which allows us to assist you. It is a firewall friendly application.

1. In order to get RTS, you must contact us [http://www.call-replay.com/Contact](http://www.call-replay.com/Contact) and let us know what problem you met. We recommend to reach us by instant messenger or email.

2. Windows Start / Programs / CallRecorder / Support / Team Viewer

Send to technical support, by email or instant messenger, your ID and Password

Wait for the connection to be established. The program will not work if the support technician was not notified to open his side.

7.4. Sending Logs to Technical Support

To enable technical support to understand the cause of a problem, the recording application keeps extensive logs of all the actions it does. To send them to us:

From application menu Help / Send Logs
1. Describe the problem as detailed as possible, including the phone numbers and IP addresses with issues.
2. Fill in the problem date correctly, or the logs will not be copied from the correct period.
3. Contact Information. Please enter your name, e-mail address and telephone number. As for the method of sending the resulting archive, we recommend using the **Upload to FTP** option.
4. Click **Submit report**.
8. Users Management

You can organize your agents, supervisors and organizational hierarchy by clicking on System / Users. This page is only available for users with “Is Admin” permission.

8.1. Adding a new department

Click on the parent department of the new department then click the Add button and choose Add Department

8.2. Adding a new user

Click on the parent department of the new user then click the Add button and choose Add User.

General Tab

User ID: account ID used for login.
PIN: short numerical password used only from the phone.
Departments: select one or more departments to which the user belongs.
By associating a phone to a user, you can easily filter the calls in "Replay Calls" Page, by selecting the respective agent node in the left-side tree. Association can be done by phone number, ip address, or mac address. We recommend you to use MACs because they are the most reliable.

Also, the phones associated to user and PIN from General tab are used as extensions and PIN in Phone Service menu in order to listen recordings or monitoring calls (see User Guide - Phone Service Menu). Filtering by MACs does not work in WAN environments, where there are routers between the recorder and phones.
Granting permissions

The values for each permission can be Inherit (keep the role permission), Allow or Deny.

Call Permissions are for:

- Replay Calls on a specific department and its sub-departments.
- Is Admin: the administrator has all the permissions and may perform any operation.
- Replay All Calls: see all the calls, regardless of user's phone settings.
- Delete Calls: may delete the calls he sees.
- Edit Tags: may define new tags or edit existing ones
- Assign Tags to Calls: may label the calls with one or more existing tags from the Replay page.
- Login and replay own calls: without this permission a user may not login.
- Export calls: may save or email the call recordings.
- Audit calls: may open the audit page.
- Assign tags to calls.
- Edit Quality Standard: may define or edit quality standards
- Fill Questionnaires: may use existing quality standards
- View quality reports
Roles

Roles are collections of permissions similar to Windows user groups. The application has 3 built-in roles: Administrators Group, Supervisors Group and Agents Group.

<table>
<thead>
<tr>
<th>Supervisors</th>
<th>can replay his own calls and the calls in all the departments and their sub-departments.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Agents</td>
<td>can only access his own calls</td>
</tr>
</tbody>
</table>

An administrator can also configure the application so that a manager can only replay calls, with the exception that a manager can view all calls within the filter set. The administrator account settings are made in application web site.
Adding or editing a role

Press the Add button and set permissions for this new generic role. The permissions for existing role can be changed – press Edit button and set the values accordingly.

8.3. Authentication Methods

The application allows two authentication ways: DB authentication and Active Directory/LDAP authentication. AD authentication is used to verify the password against a Windows domain controller. The user and its permissions must still exist in the call recorder database.
8.4. Phone Directory

This feature allows to identify the caller and destination phone numbers of a call when caller ID is not available. You can organize your contacts hierarchically on organizations and sub-departments by clicking on System / Phone Directory.

![Phone Directory Screenshot](image)

**Adding External Organization / Department**

Press Add button and choose Add Directory for adding new directory.

**Adding a new external contact**

Press Add button and choose Add User for adding new contact. Each contact can have one or more phones. After adding an external contact, each call with that contact will have it colored in blue, with a tool-tip when hovering the cursor over the contact name.

![Edit User Screenshot](image)
8.4. Change Password

Each user that has permission to log in to application site can change his password and pin by clicking Session / Change Password.

He can change only password, only pin or both by leaving blank the undesired field.
9. Email configuration

1. Enter your SMTP server credentials (SMTP server, Authentication User, password).
2. The Admin Email Password is also used for critical alerts.
3. Click Save.
10. HQ / Branch Replication

This feature allows copying or moving calls to another computer for various purposes: safety, centralized management, high availability recording etc.

First, you need to configure the Headquarter (HQ) to accept incoming connections. For this, check 'Act as HQ Controller', then enter a password for connection and hit 'Save' button. Please note down the 'HQ Address' value, you will need it to configure branches.

To configure one other computer as Branch, you need to go to the 'Branch' tab:

You can replicate calls to multiple Headquarters. First click 'Add HQ' then enter the Headquarter address and port (the one you previously noted down). Also, enter the password you've set on headquarter computer.

After this, you will be asked if you want to replicate all existing calls or only the calls made from now on.
If the connection is working, you should see on headquarter that a new entry has been added (after clicking ‘Refresh’) for the newly configured branch.

You may now configure the incoming replication type. Call information only can be used for centralized management. These calls can be played back only when the branch is running. If replication is done with audio files, you can playback them directly from the headquarter. In this case, you can also choose to delete the successfully replicated call from branch.
11. Recording Policy

This is the place for you to describe which calls will be recorded and how to prioritize your licensed recording resources. It is accessible from the Capture / Recording Policy menu. There are 2 licensing resource allocation types: Reserved Channels, Dynamic (Unnamed) Channels. Rules from each tab are evaluated in order and if a match is done the next rules are not evaluated. Each licensing allocation type can be triggered automatically or manually (On-Demand).

11.1. Reserved Channels

Reserved channels tab contains the list of phones for which the recording will start automatically. Each such guaranteed recording resource will consume one licensing channel. It is recommended that you add reserved phones by MAC address. To disable recording of phones which are not in the reserved list, set Automatic Channels to 0 in the Dynamic Channels page.

11.2. Dynamic Channels

Dynamic (Unnamed) Channels tab allows you to set the number of channels used for recording First Come First Served recording channel allocation.
11.3. Advanced tab

Record unlicensed calls: [ ]
Record inbound calls*: [ ]
Record outbound calls*: [ ]
Always record parked calls: [ ] (even if 'Record outbound calls' is disabled)
Always record transferred calls: [ ] (even if 'Record outbound calls' is disabled)
Record internal calls*: [ ]

Call demand retention: [ ]
Recording channel: [ ]
Monitoring port: [ ]
Lookup call party name: [ ]
Use effective called party number: [ ] (disregard Hunt Number, forced recording only)
Remove empty calls: [ ]

* you must configure the PSTN Gateways in PRI Configuration to get Inbound / Outbound / Internal identification
12. Silence compression (Voice Activity Detection)

Accessible from Capture / Silence Compression menu entry.

Voice Activity Detection is used with IP Trade Turrets to trigger call recording based on the voice signal intensity. It can also be used with Radio Gateways and other analog-to-IP equipment with no call signalling.

VAD Sensitivity: the level of voice activation detection from which the signal is not considered silence. The default value is Normal (40dB). Other values are Low (30 dB, noisy environment) and High (50 dB, Catch every breath).

## Silence Compression Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>VAD Sensitivity</td>
<td>Normal (40 dB)</td>
</tr>
<tr>
<td>Silence Compression</td>
<td>Keep silence within recordings</td>
</tr>
<tr>
<td>VAD separation interval</td>
<td>0.5 seconds</td>
</tr>
<tr>
<td>Recording prolog duration</td>
<td>1 seconds</td>
</tr>
<tr>
<td>Recording prolog threshold</td>
<td>75%</td>
</tr>
<tr>
<td>Silence before stopping VAD calls</td>
<td>300 seconds</td>
</tr>
<tr>
<td>Maximum VAD call duration</td>
<td>900 seconds</td>
</tr>
</tbody>
</table>

Silence Compression: keep or not to-keep the silence within recordings

VAD separation interval: it is enabled when silence is removed within recordings. This is the length of the silence inserted between active signals.

Recording prolog duration: it means the length of the interval recorded before the moment of active signal.

Silence before stopping VAD calls: after specified value of silence the recorded call will be stopped.
13. Storage Management

Recording Path Configuration

**Storage Volume Page** allows to set the folder where the calls are stored. You can add new folders by clicking Add New Volume button. You can have multiple volumes that store calls on separate drives, which will be used in a round-robin order. 

*To move calls from one folder to another:*

- Edit the volume and change its path to the new destination path.
- Move the files of the old volume to the new volume path using Windows Explorer.

Storage Quota - calls cannot exceed this value (0 means unlimited).

Reserved Space - this is the reserved storage space for CallReplay; that is, when Used Space + Drive Free Space becomes lower than Reserved Space, a warning e-mail will be sent to the administrator.

Deleted - a deleted volume is not removed when it contains calls, will be just marked as deleted and no other calls will be stored on it. You can manually delete a volume without calls.

File Renaming

**File Name Templates Page** allows the administrator to change the template used to generate the file names for storing recorded calls.

The file names template is a string containing macros that expand to values related to the call like the call ID or caller number. The list of macro names that can be used is displayed in the "Fields list" table as can be seen in the image. Click on a macro name from the table to have the macro included at the end of the current file names template. You can also type any macro name from the table in the File Name Template field, surrounded by barces { and }, to include the macro in the file name template string.

The Example field (below the File Name Template field) shows how a file name would look like using the currently displayed template string. Also watch the Example field for errors like a wrong macro name being typed in the template string.

A new file name template takes effect after the Save button is pressed, and the new template will be used for all calls recorded thereafter. To also apply the new template to all the old files, that is to rename all previous files using the new template, use the "Rename existing files?" radio group and select the option "On". For this case, you can use the "Call count" field as an estimate of the number of calls that would have to be renamed with this option.

After saving a new template with the option to rename existing files, you can see the rename progress in the Call count field, in the format "remaning / total", like for example "120 / 550".

The "Cancel" button restores the current template string into the File Name Template field, and discards any
changes you might have typed in the input box.

<table>
<thead>
<tr>
<th>File name templates</th>
</tr>
</thead>
<tbody>
<tr>
<td>File Name Template:</td>
</tr>
<tr>
<td>(TenantName ?YYYY)MMDD(YYYY)MMDD(YYMM)(mm)CallerNumber(StorageFormat)GlobalID</td>
</tr>
<tr>
<td>Example:</td>
</tr>
<tr>
<td>(storage-volume-path)/currentTenant2014040202014-04-02-02-34-0243331-0133342.sps938ba30-a02-11e3-a5c0-0800200c9a66</td>
</tr>
</tbody>
</table>

Resume existing file?  
- On  
- Off

Call count: 4906390

<table>
<thead>
<tr>
<th>Fields list</th>
</tr>
</thead>
<tbody>
<tr>
<td>Field name</td>
</tr>
<tr>
<td>YYYY</td>
</tr>
<tr>
<td>Month</td>
</tr>
<tr>
<td>MMM</td>
</tr>
<tr>
<td>MM</td>
</tr>
<tr>
<td>Week</td>
</tr>
<tr>
<td>Weekday</td>
</tr>
<tr>
<td>Day</td>
</tr>
<tr>
<td>ShortDay</td>
</tr>
<tr>
<td>DD</td>
</tr>
<tr>
<td>HH</td>
</tr>
<tr>
<td>MM</td>
</tr>
</tbody>
</table>
Recording Encryption

Introduction

We value your privacy, so we introduced an OpenPGP public key encryption of recorded calls. The encrypted file conforms with OpenPGP standard (RFC 4880). The encryption is done using AES 256. The generated certificate is a 1024-bit RSA key OpenPGP.

Setting up call encryption

1. Configure CallReplay to encrypt calls.

   a) Download Java security libraries required for encryption from: [http://www.oracle.com/technetwork/java/javase/downloads/jce-7-download-432124.html](http://www.oracle.com/technetwork/java/javase/downloads/jce-7-download-432124.html). Unzip the downloaded file and copy the two jars in the folder Java\lib\security under Call Replay installation folder. For a standard installation this is: 'C:\Program Files (x86)\CallReplay\System\Java\lib\security'.

   b) Login as administrator on CallReplay and go to Storage > Recording Encryption. You can generate an OpenPGP private/public key pair using 'Generate' button or use the 'Upload' button from the same page and select your OpenPGP public key file. If successful, you will see some of the details of the public key certificate. For the purpose of call encryption, the certificate does not need to be signed. When you generate a certificate, after entering a name and a password, you will be prompted to save the private key to your computer. Make sure to save the private key in a known location and also to remember the password. You'll need them later for call decryption.

   After generating a certificate select the 'Encrypt Calls' option. When selecting this option you'll be asked whether or not you want to encrypt existing calls. If you choose 'Yes' all existing calls with storage format WAV or SPX (Storage > Advanced > Recording Storage Format) will be automatically encrypted. If you choose 'No', old calls will be left as they are, but encryption will still be applied each time you choose to Play/Save one of them. This also applies for calls (either old or new) with storage format RTP. Configure computer for playing encrypted calls.

   You can check if calls are encrypted in 'Call Details' > File Name. Encrypted calls will have the extension PGP.

   Also in the same page you can view the encryption details in field: Encoding Format.
2. Configure computer for playing encrypted calls

The only way to listen to your encrypted calls is to download them to a computer, decrypt and listen to them locally using your favourite WAV/SPX player. This process can be automated by doing the following steps:

a) Download and install:

- GPG4Win (http://www.gpg4win.org/download.html)
- 7-Zip (http://www.7-zip.org/download.html)
- CallReplayPlayPGPSsetup.exe from CallReplay site : Playback > Desktop Encryption Helper

b) Import the private key saved at 1.b) to GPG4Win.

c) Set the default player for audio files, appropriate for the format used to save the calls (Storage > Advanced > Recording Storage Format). In case the Storage Format is SPX or RTP, then you'll need a SPX player. This can be downloaded from CallReplay site, Playback > Desktop SPX Player.

When you click on Play, you'll be asked if you want to Open or Save the file. If you choose Open, the encrypted file is downloaded to your computer and is automatically decrypted and played with the default player. The software will ask you for the password set at 1.b).

If you choose to save the file you can play it in the same way by double-click the PGP file. This will create a folder with the name composed by the name of the PGP file followed by the current time, that contains two files: the audio file itself and a text file in XML format with extension “call” where you may find informations related to the call (phone
numbers, IP address,...).
14. Retention Policy

To modify policy click on Storage / Retention Policy.

In this page you can modify policy rule. By Default, the rule records all calls and keeps calls until there is not enough free disk space.

Retention

- Keep for period (x days or x hours or x minutes)
- Keep Until Backup – the call will be kept until the first backup.
- Keep until space is required - The call will be kept until there is no more space on disk, then the oldest calls will be deleted to make space for new calls.
15. Audit

Audit Page allows to see who listened a call or who emailed a call. You can filter by auditor or / and by phone number. This option is accessible through Recordings / Audit

15.1. Email Notifications (Recordings / Email Notifications)

This page allows to configure the rule for sending emails when a specific call occurs.

The application will send emails when call starts or ends for a specified caller and called according to rules defined.

New Email Notification Rule

- Caller Party: admin
- Called Party: john
- Also reverse: Yes
- Send email when call starts
- Emails: admin@mail.org

[Image of Audit Page]

[Image of New Email Notification Rule]

[Image of CallRecorder Administration Panel]
16. Backup (Backup / Restore)

16.1. Backup

To Backup call recordings to a local folder or a windows network share, use the Backup page.

16.1.1. General Tab

In the above image, you can see the General tab, which contains usual backup options.

- The **backup label** is always appended to the backup root directory, and represents the backup folder name.
- The **full path** is the location where the backup will be done. Pressing the "..." button, will focus the backup root directory to be changed.
- The **maximum backup size** is a value that limits total call recordings backup size (example: to fit some external disk)

![Backup Options](image)
The archive type can have one of the following values:
- Incremental - only calls that haven’t been backed up will be included
- Full - all calls will be included (other filters applied)

After backup, you can choose three options:
- Don’t Remove - calls will remain in the database and on disk as they are
- Remove File, Keep Call Info - will remove the recording files, but will keep the calls in the database, and will not allow you to listen them
- Remove File and Call Info - will remove the recording files from disk, and remove all calls information from database

The free space from the backup root directory is shown on the right

16.1.2. Advanced Tab

This tab includes additional options for backup.

The backups directory is the backup root directory, where a new folder will be created with the same name as the backup label.

You can enter a local folder location here, or a windows network share with the following format: \<server>\<path>

When a windows network share is inputted, additional fields will be shown for entering domain, username and password.
The **backup period** has three choices:

- **Any** calls will be included
- **Between** - only calls that have the starting date between the two values will be included. If one date is empty, then only one date will be used as the interval margin. (example: start date is empty, end date is non-empty, then only calls with a start date below entered end date will be considered for backup).

  **Period:** Between **12/17/2013** and **04/26/2014**

- **Older than** - only calls that have the starting date older than input value will be included

  **Period:** Older than **3** Months

You can choose which calls to be backed up by **departments**, clicking the departments combo, and a pop-up will appear:
16.1.3 Status Tab

The status tab is visible only when the backup process starts.
In the progress bar, processed calls and remaining calls count are shown.

The text area below shows errors occurred during backup.

### 16.2. Backup History

To view a history of backups made and their details, use the Backup History page:
The left grid shows all backups made. You can sort entries by clicking any column header.

The right panel shows selected backup details.

### 16.3. Restore

To Restore a backup made in a windows network share or local folder, use the Restore page.

#### 16.3.1. Options

To view backups, select a backup root folder or a backup folder from which to restore calls.

You can also select a location from a windows network share (following the pattern `\<server>\<path>` ) and login information will be shown:
If no login information is inputted, then the default user GUEST will be used.

You can sort backup entries in the left grid by clicking on a column header.

In the right panel, backup details of the selected backup are shown (same as backup history).

The refresh button located right of the input directory helps rescan for backups.

You can select either to move files from the backup or to copy them.

- If move files is selected, the backup files are removed during restore process.
- If copy files is selected, the backup files remain intact.

To start a restore operation you can:

- click restore button in the last column of the grid, which restores the backup form that row

- click restore button from the details panel to restore selected backup

16.3.2. Status

Once the restore operation has started, the status tab becomes visible:
If any error occur, it will be shown in the text area located below the progress bar.
17. Manually Configuring the IP Phone Service

17.1. Phone Service

(This chapter has been obsoleted by the Configure Cisco CM wizard in version 6). Please use the Forked Recording Wizard in the Capture / PBXs page.

Observation: Instructions for setting up a Cisco CallManager Express phone service can be found on Cisco’s site:


The IP phone service is a component of our application that allows users with a primary extension to listen to their calls from their Cisco Phone or to demand recording of calls. By accessing the System/Watchdog menu, you can specify the authentication parameters used with the Cisco Phone XML, and other values that affect the way that the service works:

- **Cisco User** - The user name used for authentication with the Cisco Phone XML.
- **Cisco Password** - The password used for authentication with the Cisco Phone XML. You don't need to specify the password each time that you make an update in the configuration page, but only when you need to change the existing password.

![Phone Service Configuration](image)

17.2. Application Phone XML User

This user is required for the application phone services to function properly. There is only one application Phone XML User, and it is different from the site users. You may choose any user you want, but we recommend creating a new special user that nobody else uses.
Create a new Cisco User or select an existing one
1. Go to the CallManager Administration Site
2. Go to User/Global Directory
3. Click "Add a New User". For more details about adding a new user, please consult the CallManager Help.

20.3. Associate all devices with the desired user

*Make sure that this user has all the devices associated to him.*

Go to the CallManager Administration Site
1. Go to User/Global Directory, then click "Search"
2. Select or create the user you want to use with Cisco Phone XML. For example "CallRecorder"
3. In the User Configuration page, please click "Device Association"
4. Now you have to associate all the phones through which you want to access the application and/or Call Monitoring phone services with this user. If you want to associate all devices, do the following:
5. Press "Select Devices" (leaving the search field empty)
6. Select "Check All in Search"
7. Click "Update Selected".

17.4. Setting up the authentication

- Go to System/IP Phone Service
- In the "IP Phone Service" page enter the following values:
  - In the "Cisco User" field, enter the name of the user that you associated all your phones with (see the previous step)
  - In the "Cisco Password" field enter the password of that user
- Press "Save"

To make the application service accessible on your Cisco IP phones, you have to go through the following steps:

17.5. Add a new service

**Adding new service**
1. Go to CallManager Administration / Device / Device Settings / Phones Services
2. Press "Add New" button
3. Set "Service Name" to **Call Recorder**
4. Service Category must be XML Service and Service Type Standard IP Phone Service
5. Set "Service URL" to [http://CallRecorderServer(:port)/CallRecorder/phoneService](http://CallRecorderServer(:port)/CallRecorder/phoneService)
6. Click Save
   - For Demand recording the URL is [http://CallRecorderServer(:port)/CallRecorder/phoneService/onDemandPhoneDemandThisCall](http://CallRecorderServer(:port)/CallRecorder/phoneService/onDemandPhoneDemandThisCall)

17.6. Assign the service to the phones you want

*For large number of phones this is best done using Device Profiles.*

Using the Call Manager Administration site:
1. Go to the CallManager Admin/Device/Phone
2. For each phone having access to CallRecorder, do the following:
3. Go to that phone's configuration page
4. Click "Subscribe/Unsubscribe Services"
5. In the available services list select CallRecorder, then click "Continue"
6. Click "Subscribe"
7. Click "Update"
8. Restart the phones so that they can read the new configuration (only if you had to change the URL to the value we specified)
   a. Go to CallManager Admin/Device/Phone

For each page "Select all", click "Reset", then "Restart".

17.7. Application Service Control (Watchdog)

By accessing the System/Watchdog menu, you can set the automatically reboot moment of the application service.

![Watchdog Configuration](image)

**Watchdog Configuration**

- **Total Memory:** 491.00 MB
- **Current Memory Usage:** 26.38 MB
- **Out of Memory Threshold:** 90 %
- **Automatically Restart Service:** Never at 00:00 (24h) [Save]

[Dashboard Image]
18. Configuring Speed Dial

Steps to follow to configure a speed dial button for using a phone service

1. Define a phone service (Call Manager: Device> Device Settings> Phone Services), as described in Chapter 19, for this URL:

   [URL]

2. Create a new phone button template for your phones (Call Manager: Device> Device Settings> Phone Button Template) and select "Service URL" for button which will be used as a speed dial for our phone service.

3. Go to Device> Phone and select new defined template on phones
4. Subscribe the phone to the phone service
5. The last step is to assign the new subscribed phone service to the phone button (configured on step 2). After phone reset, the speed dial is ready for use.
19. CallRecorder Firewall Ports

Ports opened on the CallRecorder server

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Port</th>
<th>Service Description</th>
<th>Required</th>
</tr>
</thead>
<tbody>
<tr>
<td>TCP</td>
<td>843</td>
<td>Flash policy server</td>
<td>No</td>
</tr>
<tr>
<td>UDP</td>
<td>1026</td>
<td>Database</td>
<td>No</td>
</tr>
<tr>
<td>TCP/UDP</td>
<td>5060</td>
<td>SIP default port</td>
<td>Yes</td>
</tr>
<tr>
<td>TCP</td>
<td>5432</td>
<td>Database</td>
<td>Yes</td>
</tr>
<tr>
<td>TCP</td>
<td>8080</td>
<td>Jetty web server</td>
<td>Yes</td>
</tr>
<tr>
<td>TCP</td>
<td>8081</td>
<td>Monitoring calls</td>
<td>No</td>
</tr>
<tr>
<td>TCP</td>
<td>8079</td>
<td>Software update</td>
<td>No</td>
</tr>
<tr>
<td>TCP</td>
<td>9000</td>
<td>Licensing</td>
<td>No</td>
</tr>
<tr>
<td>TCP</td>
<td>9853</td>
<td>Replication</td>
<td>No</td>
</tr>
<tr>
<td>UDP</td>
<td>25000-30000</td>
<td>RTP port range (even values)</td>
<td>Yes</td>
</tr>
</tbody>
</table>

CallManager Ports Required by Forked Recording

This ports must be opened on the CM server, or Forked will not work

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Port</th>
<th>Service Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>TCP</td>
<td>2748</td>
<td>CTI</td>
</tr>
<tr>
<td>TCP</td>
<td>2749</td>
<td>JTAPI</td>
</tr>
<tr>
<td>TCP</td>
<td>2789</td>
<td>SIP default port</td>
</tr>
<tr>
<td>TCP</td>
<td>8443</td>
<td>AXL</td>
</tr>
<tr>
<td>TCP</td>
<td>443</td>
<td>HTTPS</td>
</tr>
<tr>
<td>UDP/TCP</td>
<td>5060</td>
<td>SIP</td>
</tr>
</tbody>
</table>

20. Localization

Requirements

For creating a custom localization of CallRecorder you need at least version 7.1.10 of application installed using a full setup.

Creating/editing CallReplay localization

CallReplay CallRecoer has two major components who need to be translated separately. One component is recorder service and other is site administration.

First thing which must be done is to create a copy of entire Translations folder into CallReplay data folder. Translations folder can be found in installation path, usually C:\Program Files\CallReplay\System\Translations. Data folder is the folder where the database is kept, default value is C:\CallReplay.

Once we have new Translations folder we can start to create/edit translations. All customizations must be done in this new Translations folder otherwise will be lost after first running of CallReplay Software Update or after a full setup.

The tools used for translation are "translate_site.cmd" and "translate_server.cmd".

Create a new translation for CallRecorder site

Go to the new Translations folder created using indications from previous paragraph.

Duble-click on "translate_site.cmd" file. If this is the first time when the translation tool is used you must select a language for translation tool interface:

After language selection the main window of translation tools is shown:
Press the add language button:

Select desired language from the list:
The result is a new Language node:

Now select the new language and translate every key from the master file to the new language. Next picture show how to do translate "All rights are reserved" to Korean.
The untranslated keys are shown with blue color. So it is easy to know which keys are translated and which not.

For saving the new language press "CTRL+S" which is a shortcut for File>Save option and a new file having name "site_xx.properties" will be created where xx is the language code.

**Create a new localization for CallRecorder service**

Go to new Translations folder and double-click on "translate_server.cmd". The same tool as for translating site will be shown excepting that the keys are for CallRecorder service.

Create a new localization for CallRecorder service following the same steps as for CallRecorder site. The name of the new translation will be "server_xx.properties" where xx is the language code.

**Activate a new localization**

A new localization becomes active after including in file "locale.properties" the line: xx=Language and restarting of CallRecorder service.

e.g.: ko=

**Editing an existing localization for CallRecorder site and service**

Run the translation tool ("translate_site.cmd" for site and "translate_server.cmd" for service ), select the language and edit the keys accordingly. Save the changes (CTRL+S or File > Save ).
CallReplay Cloud Server

Introduction

The Cloud Server is a feature of CallReplay which allows you to record calls and upload them on a properly configured CallReplay Cloud Server.

The Cloud Server can be used as a Replication HQ for other CallReplay servers or mobile recorders.

You can make use of this feature in two ways:

1. Uploading to the public CallReplay Cloud Server located at [http://cloud.call-replay.com](http://cloud.call-replay.com);
2. Configure your own CallReplay Cloud Server - this way you have all features of CallReplay Call Recorder.

Supported mobile phones: Android 2.3.3 or higher; other operating systems may be supported in the future.

Configuring your own CallReplay Cloud Server

First of all, you should acquire a license for this. The license should contain how many tenants you need (i.e. cloud accounts) and how many phones and mobiles you want to record and upload calls.

Upload this license in your CallReplay installation using System -> Licensing.

Add Tenants

then go to System -> Tenants page. You can add your tenants here.

First enter the company name and fill the TLD field with your domain name / website if any.

You can assign a number of Replication Branches, if you want to make this tenant a Replication HQ. In this case,
you should allocate a number of channels to those branches and assign them individually from HQ/Branch Licensing page. That is, if you want branches A, B and C to upload calls to this tenant, write 3 in 'Branch' edit box. Then, if you want A to have 5 channels, B and C have 10, write 25 in 'Channels' edit box, then later go to HQ/Branch Licensing and assign each of them the corresponding channel number. Here you will introduce the total number of channels allocated to the group of branches associated with this tenant.

In 'Mobile Phones' you will write the total number of mobile phones that can upload calls to this tenant (cloud account).

'Storage Quota' - how much of storage space will be allocated to this tenant from the total storage pool. Older records will be deleted when the quota is reached. Enter 0 if you want to use global cleaning.

'Active' - use this checkbox to activate / deactivate the tenant.

'Validity' - if you want to automatically deactivate the tenant after a period of time.

After pressing 'OK' button, a new tenant will be generated and you can see it in the main list. There you can find the generated Tenant ID used for tenant identification.

**Add Tenant Admin**

You may create multiple user accounts for each tenant, at least one of them should have administrative rights. You can quickly create such an administrative account using 'Add Admin' button from 'Tenants' page.

A password will be randomly generated for the tenant administrator and it will be sent to the specified e-mail address.

**Public Tenant Creation**

This feature is available only on public CallReplay Cloud Server located at [http://cloud.call-replay.com](http://cloud.call-replay.com). Anybody can create a trial tenant for itself. This trial tenant will expire after 1 month and its storage quota is limited to 200MB.

To convert a trial tenant to a permanent tentant, please contact sales or support.

**Tenant Login**

Upon creation of the its account, the administrator will receive an e-mail containing the login credentials. In order to access the Cloud Server, the administrator (and users created subsequently) must enter the Tenant ID along with the login name and password.

After login, you have access to all recorded and uploaded calls. You can playback them (if not encrypted), e-mail them and so on.
Android Recorder

Introduction

CallReplay Recorder for Android allows you to record calls you make on your Android phones. Since the space available to store these recordings can be very limited, you can upload them on CallReplay Cloud Server.

For this, you need to have an account on a CallReplay Cloud Server. If you intend to use the public CallReplay Cloud Server located at http://cloud.call-replay.com please follow the next step.

If you want to use a custom installation of CallReplay Cloud Server, please skip to the 'Configuring phone to upload calls' section.

The minimum supported version is Android 2.3.3.

For the best experience with Cloud Server, you need a Flash enabled desktop/laptop/tablet browser. The cloud application is not optimized for mobile phone screens.

Note: in this manual, the terms 'tenant', 'tenant account', 'cloud account', 'organization' all refers to the cloud space created in CallReplay for your organization. 'Organization ID' is the generated ID of this cloud space. The terms 'administrator', 'administrative user' and 'administrative account' refers to the person who has the login credentials needed to manage your tenant account (received in the e-mail specified at tenant creation). The administrator could create more administrative users for the same tenant, and also more non-administrative users.

Creating a public Cloud Server tenant account

Using a Flash enabled browser, go to http://call-replay.com/go/register.
You need to enter at least a valid e-mail, your name and your company name. The trial account created this way is valid for 1 month and have 200 MB of disk space at your disposal.

You can use the same account for multiple phones, however, the trial account is valid for maximum 5 phones.

If you want to extend the validity of the account or change the available disk space or phone numbers, please contact sales or support ( see http://call-replay.com ).

You will receive an e-mail to the address provided, containing login credentials for administrative account (administrator) of this tenant

Note: if you've wondered what is the 'Request Cloud Account' item from 'Upload Server' section of settings page, it has been added for people who do not read the manual. If we've shown the link in the Android Call Recorder, most people will tend to click it and expect to work on their mobile phone. Since it doesn't, and we do not want them to type the whole address, we send them an e-mail which they can read on another computer and click the link.

Creating a tenant account on a custom installation of CallReplay Cloud Server

Please see this manual page.
Configuring phone to upload calls

Upon registering a new tenant (cloud account), you will receive an e-mail containing credentials required to access the cloud server.

Please go to ‘Settings’ section and click on ‘Upload Server’. You will see a page like this:

‘Upload Server’ - cloud.call-replay.com for public CallReplay Cloud Server, your server name / IP for custom installation of CallReplay.

‘Organization ID’ - this is your tenant account ID; it is named ‘Organization ID’ to avoid confusion with administrator ID. A since you can upload calls from multiple phones on the same account.

‘Password’ - this is the upload password, not your administrator password. To can change this password, login to your tenant administrator account, and go to ‘Headquarters / Branch Replication’. Here you can find the 'connection password' on ‘Headquarters tab’.

‘Use Wi-Fi’ - to automatically upload calls whenever a Wi-Fi connection is available.

‘Use mobile connections’ - useful when a Wi-Fi connection is not available. The upload on the mobile connection can be slower and might be subject to additional fees from your mobile carrier.

‘On roaming’ - check this if you want to upload calls when you travel to other countries and want to upload calls.
Warning: this can be costly.

'Test Upload' - click this to see if all is configured correctly.

'Delete after Upload' - check this if you want to automatically delete uploaded records from your phone - this way you make space to record other calls.

'Encryption certificate' - see this page.
Call Encryption

Introduction

We value your privacy, so we introduced an OpenPGP public key encryption of recorded calls. If you do not have an OpenPGP public / private key pair, you can generate them using a 3rd party software which you can find freely on Internet (see notes on the bottom of the page).

Setting up call encryption

Login as administrator on CallReplay Cloud Server and go to 'Storage' / 'Recording Encryption'.

Click the 'Upload' button and select your OpenPGP public key file. If successful, you will see some of the details of the public key certificate. For the purpose of call encryption, the certificate does not need to be signed.

Upon the next connection of your mobile phones to your tenant, the certificate will be sent to all of them and all subsequent calls will be encrypted using this certificate.

You can change the certificate any time you want, and only subsequent calls will be encrypted with the new certificate, the older ones will remain as they were (unencrypted or encrypted with older certificates).

Notes:

1. Uploading only the public key gives you a great deal of privacy: in case of mobile phone theft, the calls could not be listened. The drawback is that neither you can listen to your calls on the mobile phone.
2. In case you've wondered if someone could listen to your calls uploaded to CallReplay Cloud Server, this way you can be sure nobody could. The drawback is the same: neither you can listen to your calls online.

The only way to listen to your encrypted calls is to download them to a computer, decrypt them with your PGP capable software of your choice. The software will ask you for the private key, will decrypt your files and you can listen to them using your favorite WAV/3GPP player.

Quick setup for call encryption

You can also generate an OpenPGP private/public key pair using 'Generate' button from the same page. After entering a name and a password, you will be prompted to save the private key to your computer.

The generated public key will be sent to your mobile phones and calls will be encrypted using it. However, the private key will not be kept on our server for security reasons, so you should take proper care of the saved key. If you lose it, your calls cannot be decrypted.

Mobile phone encryption configuration

For the encryption to work, after you generated or uploaded a public key on CallReplay Cloud Server, you need to connect your mobile to the server. Make sure you have Internet connection then either go to 'Status' page and click 'Upload Now'.

Then go to the 'Settings' page, 'Upload Server' section and you should see the new info on 'Encryption Certificate' item.

Note: in case you have used multiple certificates over time, for each encrypted call you will see the certificate details on the call details page; this way you can identify the private key required to decrypt the recorded call.

Notes
The encrypted file conforms with OpenPGP standard (RFC 4880). The encryption is done using AES 256.

The generated certificate is a 1024-bit RSA key OpenPGP.

If you do not have an OpenPGP certificate and prefer creating one yourself, you can use GPG4Win, which is a 3rd party OpenPGP compliant free software for Windows: http://www.gpg4win.org.
## Tested Android Handsets

<table>
<thead>
<tr>
<th>Date</th>
<th>Model</th>
<th>Notes</th>
<th>Codec</th>
<th>Duration</th>
<th>Callant Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>6-Sep-2013</td>
<td>Galaxy Young GT631 2</td>
<td>Înregistrează numai cu AMR în ambele direcții la o calitate decentă cu mici distorsiuni.</td>
<td>Înregistrează numai cu AMR și WAVE ca microfon extern.</td>
<td>4.1.2</td>
<td>N/A</td>
</tr>
<tr>
<td>2013</td>
<td>Galaxy S3 Mini GT-I8190</td>
<td>N/A</td>
<td>Înregistrează numai WAVE ca un microfon extern.</td>
<td>4.1.2</td>
<td>N/A</td>
</tr>
<tr>
<td>Data</td>
<td>Dispozitiv</td>
<td>Rezultate</td>
<td>Rezultate</td>
<td>Observații</td>
<td></td>
</tr>
<tr>
<td>--------------</td>
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<td>-----------</td>
<td>-----------</td>
<td>---------------------</td>
<td></td>
</tr>
<tr>
<td>10-Sep-2013</td>
<td>LG 2</td>
<td>N/A</td>
<td>N/A</td>
<td>Înregistrare WAVE și cu o reducere sensibilă de calitate AMR (înregistrarea ambelor canale la fel ca VoiceCall-ul)</td>
<td>4.1.2</td>
</tr>
<tr>
<td>2013</td>
<td>Samsung Galaxy Note 2</td>
<td>WAV + AMR în ambele directii</td>
<td>4.1.2</td>
<td>-</td>
<td>Ok</td>
</tr>
</tbody>
</table>
Release Notes

- Old Versions
- Version 6.x
- Version 6.5
- Version 7.x
- Version 8.0
Old Versions

5.3.05 (11-Feb-2010)
- NEW: improved database performance, decreased size
- NEW: improved Flash loading time
- NEW: Audit permission
- FIX: many cosmetic fixes

5.2.00 (07-Sep-2009)
- NEW: improved playback performance with embedded Speex / WAV player for maximum playback speed (15x playback startup speed).
- NEW: added HTTP access to calls for easier integration with 3rd party apps
- FIX: click on player slider works now

5.1.05-07 (09-Aug-2009)
- CRITICAL: service was stopping on power events
- CRITICAL: storage manager was deleting calls when large "Min Free Space" values where used
- FIX: out-of-sync conversations
- FIX: added call_id db indexes to correct performance problems
- FIX: clean Temp dir at web site startup

5.1.04 (1-Jul-2009)
- NEW: control recording of missed / unlicensed / inbound / outbound / internal parked calls
- NEW: report generator with graphs and everything to compare basic agent stats. (Quality / Compare Agents page)
- NEW: search by description
- FIX: Avaya IP Office 500 (GLH)
- FIX: SIP rtp map

5.1.03 (23-Jun-2009)
- NEW: record missed/unanswered calls in DB
- 5.1.01: critical fix in licensing code, reserved phones were sometimes not recognized
- 5.1.02: phone service ip addr fix
- FIX: G.722 fix for slices != 20 ms
- FIX: RTP stream sequenceNumber reset
- FIX: Storage was only using one volume
- FIX: VAD G711 A/U mismatch
- FIX: hold + resume on a shared line

5.1.0 (29-May-2009)

Version 5.1 refines overall the 5.0 version. We heard your observations and made the changes:

- NEW: changed licensing model from per-phone to per-concurrent call. This is effectively a discount for many organizations. All existing customers can benefit of this discount by upgrading to v5.1.
- NEW: Support for NEC IP protocols (NEC SV8100, NEC SV8300, etc)
- NEW: View Active Calls list in real-time
- NEW: more permission types
- NEW: advanced search tab
- NEW: support for G.729B
- NEW: updated documentation for v5
• CHANGED: New playback system for very long calls
• CHANGED: massively improved non-stop recording
• CHANGED: Many speed improvements in the service (from 5.0.7)
• FIXED: Many critical service bugs added in 5.0.7
• FIXED: some unpleasant memory leaks
• FIXED: config bugs with database upgrade
• FIXED: config bugs when HTTP proxies are used

In total, more than 250 changesets. Get it fast from the Downloads page.

5.0.0 (03-Mar-2009) Major changes include a 100% new user interface, based on Adobe Flash, and a rethinking of the usage model, benefiting from the 4 years of existence of CallReplay.
• NEW: integrated agent scoring and call center optimization module. The evaluation questionnaires are completely customizable. And it is bundled for FREE with CallReplay!
• NEW: support for multiple storage volumes. Self-managing, of course.
• NEW: support for recording non-stop calls (24/24), for IP trading turrets.
• NEW: licensing management web site for resellers. Automated license activation. Self-service.
• NEW: call browsing by agents. Hierarchical organization model for grouping agents in departments and sub departments.
• NEW: automatic resolution of external phone numbers using an integrated phone directory.
• Many frequent operations can be done three times easier. And they were easy before.
• Recording speed significantly faster than 4.0.
• Fault isolation for better reliability.
• Support for Windows 64 bit
• Support for Cisco CallManager 7.0.

4.0.0
• Support for Avaya Communications Manager

3.3.6 (06-May-2008)
• CWR licensing support for Cisco CMs > 5.0

3.3.5 (05-Mar-2008)
• fixed memory leak with unterminated calls
• fixed timeout error when making large backups
• added call statistics in logs

3.3.4 (08-Feb-2008)
unreleased

3.3.3 (21-Nov-2007)
• one more fix for TCP RST memory leak
• greatly decreased memory requirements

3.3.2 (20-Nov-2007)
• FIX: ugly memory leak. TCP RST now correctly closes immediately the connection
• change of call center licensing to match by latest digits of phone number
• allows the supervisor to search by the LATEST digits in the IP address of the phone (eg 10.1.0.103 -> 103)

3.3.1 (05-Oct-2007)
• FIX: Corrected error messages when Qodec could not be loaded, corrected "Object instance not found"
• cosmetic: Always displays the number of phones registered to CM

3.3.0 (22-Aug-2007)
• NEW: Replication of recordings or meta-date between multisites

3.2.0 (26-Jun-2007, unreleased)
• major rewrite of Backup & Restore functionality, improved speed and reliability

3.1.9 (3-May-2007)
• FIX: caller id is now displayed correctly for the final transfer of a call with a private number (original caller->final destination)
• cosmetic: private numbers now show as "Private number" instead of empty string

3.1.8
• FIX: shared lines fix for CCM > 4.0

3.1.7
• FIX: removed mixers memory leak
• FIX: mixers statistics

3.1.6
• FIX: CallManager Express 3.4: conference calls had incomplete CallInfo packet

3.1.5
• FIX: Service.aspx was displaying error (branding).
• FIX: clicking Update in System / Configuration was showing an error (branding).

3.1.4
• FIX: SqlServer Agent was not started in branded builds

3.1.3
• FIX: Fixed VLAN support

3.1.2
• cosmetic: improved compression tracing
• cosmetic: added backgroundCompressionAscending option (registry only)
• FIX: G.729 calls counted properly, even after conversion
• FIX: Call Manager Express: disabled handling of UDP call control protocols

3.1.1
• FIX: star numbers (like *1004) were not being recorded

3.1.0 (March 05, 2007)
• NEW: Added "Force G.729 Recompression" option in Configuration
• NEW: Backup & Restore
• NEW: PreConfiguration automatically enables weekly reboot
• FIX: fixed folder creation bug entered in 3.0.4
3.0.13
- NEW: Updated Qodec to 0.99
- FIX: PreConfiguration now displays G.729 computer id is now correct
- FIX: licenses escape strings and can accept &

3.0.12
- FIX: CM v5 support incorrectly disabled in v3.0.11
- FIX: "file does not exist" error added in previous commit
- FIX: "Must specify valid information for passing in the string".
- FIX: sometimes CallDetails loading very slowly. File URL was checked remotely for existence.

3.0.11
- FIX: db already exists (FolderDb.cs, WelcomeWizardPage.cs)
- FIX: added memory statistics

3.0.9 and 3.0.10
- FIX: RTP stream overflow check. Also an exception in RTPStream will no longer crash the program forever
- FIX: On demand
- FIX: phone detection in phone service was done by IP, so it was incorrect in presence of DHCP. Now phone service opens directly the extension input page
- FIX: activated printLeases for support
- FIX: small db fix
- FIX: shortcut creation fix
- FIX: cosmetic: added tracing for policy check

3.0.8
- FIX: various branding fixes
- FIX: LogPacker now correctly recurses subdirectories in the calls file list.

3.0.6
- FIX: critical fix for CallInfoV2 support (CM 4.2 and CM 5.0)
- FIX: corrected bug with sql scripts upgrades

3.0.5
- FIX: calls from private numbers were not recorded

3.0.4
- FIX: bug with licensing by MAC address

3.0.2 and 3.0.3
- FIX: parser fix for CallManager Express: rtp dispatched as skinny
- FIX: Improved skinny tracing

3.0.0 (Dec 13, 2006)
- NEW: Reengineered recording engine, with multiprotocol support
- NEW: Support for Cisco CallManager 4.2 and 5.0 (Skinny 8.0)
- NEW: configurable site host name
- FIX: Support for Cisco CallManager Express
Version 6.x

6.0.142

Various bugs fixes

6.0.29 (22-Mar-2011)

Massive release, with improvements across all modules and structural changes.

- NEW: SPANless recording (aka forked recording) support for Cisco UCM and phones with internal bridges (Capture / PBXs / Edit PBX).
- NEW: Live Monitoring in browser for all supported PBXs (Recordings / Replay / Play on active call).
- NEW: Email notifications triggered by specific calls (Recordings / Email Notifications).
- NEW: Automatic Software Updates at scheduled times (System / Software Updates).
- NEW: Automatic PBX detection. Recording will start as soon as the software is installed (Capture / PBXs).
- NEW: IP Phone Service and Forked Recording configuration wizard (Capture / PBXs).
- NEW: Service Memory Watchdog (System / Watchdog).
- NEW: Master / Slave: centralized licensing, centralized software update
- NEW: dedicated data folder for calls, db, logs, licenses.
- CHANGED: Backup system reworked, recordings can now be backed up using standard file system tools.
- CHANGED: user permissions system reworked
- CHANGED: licensing resource no longer aggregate, to minimize the number of licenses to be managed
- CHANGED: embedded database changed from MSDE 2000 SP4 to Postgres 9.0.3.
- CHANGED: removed .NET dependencies
- CHANGED: PreConfiguration moved in the site
- CHANGED: LogPacker moved in the site
- CHANGED: Network Interfaces moved in the site
- CHANGED: improved recording performance
- CHANGED: faster web site loading
- CHANGED: remote support tool from Remote Helpdesk to TeamViewer
- CHANGED: web server to Jetty 7.2
- CHANGED: reworked phone service
- CHANGED: major configuration changes no longer require the service to be restarted
- FIX: all fixes from 5.3 branch.
Version 6.5

- NEW: Multi-tenant hosting of multiple customers on a single server.
- NEW: Android mobile phone recording with encryption.
- NEW: Master/Slave Replication renamed to HQ/Branch Replication; now allows regional HQ servers.
- NEW: Call Recordings Encryption using Public Key Cryptography (GPG).
- NEW: HTTPS / TLSv3 connection encryption.
- NEW: Software updates using CDN or HQ server.
- NEW: Filters to displays calls with or without Quality Forms
- NEW: added Quality Report with question details
- NEW: Quick Search in Users and Phone Directory
- FIX: Forked Recording now support multiple codecs
- FIX: updated all third-party components to latest versions
Version 7.x

7.3.43 (27-Mar-2014)
- CHANGE: Tenant - expire date as date picker.
- FIX: Fixed branch licensing.
- FIX: User phones - do not allow empty lines.

7.3.42 (20-Mar-2014)
- CHANGE: reworked licensing alerts.

7.3.41 (19-Mar-2014)
- FIX: Playback - signature mismatch error.
- FIX: Delete user with evaluations.

7.3.40 (18-Mar-2014)
- NEW: Save CDR global_call_id field in callreplay database for jtapi recorded calls.

7.3.39 (07-Mar-2014)
- NEW: quick search for PBXs.
- FIX: fixed expired license message.

7.3.38 (03-Mar-2014)
- CHANGE: reworked sql session rollback.
- FIX: fixed "StartWith" policy type.
- FIX: fixed tracing replication errors.
- FIX: fixed start of call through forked recording.

7.3.37 (08-Jan-2014)
- FIX: fixed decoding mixed payload types.
- FIX: fixed database shrinking comand line on windows environment.

7.3.36 (19-Dec-2013)
- CHANGE: reworked filter management for local network traffic.

7.3.35 (18-Dec-2013)
- NEW: filter management for local network traffic.

7.3.34 (16-Dec-2013)
- NEW: transcoding calls through web site url.

7.3.33 (12-Dec-2013)
- FIX: configure database port and memory profile through Site Configuration.

7.3.32 (12-Dec-2013)
- FIX: Skinny protocol - fix recording calls for a shared line.

7.3.30 (06-Dec-2013)
- FIX: Send call by email - added missing caller/called name.
- FIX: fixed "EndWith" matching policy.

7.3.29 (02-Dec-2013)
- FIX: PBX autodetection for SPAN recording.

7.3.28 (28-Nov-2013)
- FIX: database error on starting service.
- FIX: performance issue on loading users/departments tree.

7.3.25 (22-Nov-2013)
- NEW: show volumeID on storage volume list.
- CHANGE: Call Details - limitless text for Description.
- CHANGE: Questionnaire - limitless text for Comment.
- CHANGE: Replication - generate .call files for existing calls.
- CHANGE: Licensing - allocate all channels of new licenses on default tenant.
- CHANGE: allow export a single call.
- CHANGE: Storage Volume - create temp folder inside the new defined storage volume.
- CHANGE: Replication HQ - added calls size and count field on recorder list.
- FIX: fixed SOAP error thrown by forked recording wizard (CUCM 8.0).
- FIX: Replication - fixed resumed upload.
- FIX: Software update - delete old versions.
- FIX: fixed email notification.
- FIX: Storage Volume - deny duplicate path entries.
- FIX: fixed database upgrade error caused by changing of mybatis version.
- FIX: solved sync issue while transcoding g729 rtp file.

7.3.21 (05-Nov-2013)
- FIX: solved high cpu usage.

7.3.20 (04-Nov-2013)
- FIX: fixed ports allocation for transferred calls (forked recording).

7.3.19 (31-Oct-2013)
- NEW: extended matching capability for reserved phones.
- CHANGE: configurable storage access password.

7.3.16 (28-Oct-2013)
- FIX: Reserved Phones - fixed exact match policy.

7.3.15 (28-Oct-2013)
- NEW: Audit - added new call events: close and call export.

7.3.14 (25-Oct-2013)
- FIX: Audit - fixed tracing of events.

7.3.13 (25-Oct-2013)
- FIX: other fixes for transferred calls (forked recording).
7.3.12 (18-Oct-2013)
- FIX: solved stop recording for calls longer than 15 min (forked recording).

7.3.11 (11-Oct-2013)
- FIX: SIP - record transferred calls (SPAN recording).

7.3.10 (07-Oct-2013)
- FIX: fixed hold/unhold calls (forked recording).

7.3.9 (02-Oct-2013)
- FIX: Quality - fixed “Compare Scores” report.

7.3.8 (24-Sep-2013)
- FIX: Replication - fixed replication errors caused by null values.

7.3.7 (23-Sep-2013)
- NEW: new CallReplay Transcoder installer available for download from CallReplay UI menu.
- FIX: fixed duplicated calls for CUCM 6, 7 and 8.0 (forked recording).

7.3.6 (18-Sep-2013)
- FIX: Recording policy - fixed inbound/outbound detection.
- FIX: fixed duplicated calls for cucm 8.5 (forked recording).

7.3.5 (16-Sep-2013)
- NEW: defined a new user right to close active calls

7.3.4 (11-Sep-2013)
- NEW: customize ftp credentials for sending logs
- CHANGE: deactivate pbx autodetection mechanism

Android 1.2.4
- NEW: user groups.
- NEW: recording schedule.
- NEW: maximum storage period.
- NEW: remote configuration.
- FIX: bug fixes.

7.3.3 (21-Aug-2013)
- FIX: solved database creation error on installation

7.3.2 (14-Aug-2013)
- FIX: fixed phones filter on forked recording wizard.
- FIX: fixed application page title on IE.

7.3.1 (08-Aug-2013)
- NEW: option to join storage volumes.
- CHANGE: update FlashPlayer to version 11.8
• FIX: keep the selected user in user tree during playback his calls.
• FIX: solved “MLPP Preemption must be disabled on devices” error for CUCM 8.0 (forked recording).

7.3.0 (24-Jul-2013)
• NEW: new configuration wizard for forked recording

7.2

NEW: Linux Installer

Various Changes
• Recording policy - new option to reserve phone by user name
• PBX Configuration - allow duplicate PBXs
• PBX Configuration - delete all pbxs option
• Send call by e-mail to multiple destinations

Android 1.2.0
• more responsive start/stop recording (separate thread)
• indicator for unplayable records (no audio, encrypted)
• direction recording policy
• configurable max call duration
• automatically show soft keyboard
• toast with call review
• order call groups alphabetically or by call time
• add unknown number to phone book
• added ‘call info only’ to recording policy
• favorites group
• configurable toast display while app is not visible

7.1

Licensing
• HQ/Branch licensing (removed channels allocation from Replication)
• backup existing licenses instead of delete

Phone Service
• browsing calls list
• added exit button to replay calls
• delete calls
• demand command / monitoring - show a message when the call is unlicensed
• redirect to home page after demand recording
• reload last page after playing/deleting/e-mailing a call.

Recording policy
• fixed lag on loading
• loading empty calls count on save
• remove empty calls
• quick search

View Calls
• show call score
• fixed call party names decoding according to selected charset

**Forked**
• multiple recorders connected to a single CUCM
• configuration wizard - checkboxes for selecting all phones

**Various Changes**
• Site - toolbar with shortcuts
• tenant registration - random password instead of user selected password
• transcoder - fix too much silence for ts/seq change without ssrc change
• Replication - manual retry
• View Calls - advanced tab - filter calls using '%'
• Storage Volume - reserved space
• Storage Volume - low disk space alert
• Organization tree - speed up the loading process
• Audit - aditional filters
• Setup - new parameters: /verysilent, /upgrade, /host, /port, /ip
• Site monitoring - fix broken pipe error
• SPAN Recorder - GRE( ERSPAN ) packet decoding

**Android 1.0.10 - 1.1.12**
• black list
• background version check, manually and periodically
• new design
• list of unknown numbers
• contextual help system
• app icon click goes home
• rate app dialog
• save logs as zip, to be manually sent
• progress dialog while deleting calls
• recording simultaneous calls (in the same audio file - android limitation)
• automatically close app during airplane mode
• record / pause buttons in status panel
• recording policy: automatic, on demand, no recording
• start / stop button on notification (android 4.1 and up)
• quick search calls
• access to calls/setting protected by numeric PIN

**Major Release 7.0**

**Forked Recording**
• NEW: support for forked recording redundancy using multiple servers; works best on CM >= 8.
• can now use redundant JTapi connections to clustered CMs
• support for multiple CM versions at the same time
• control of each party notification tones and volume
• refactored to insure best CM compatibility across all existing Cisco CM versions
• support for Extension Mobility Device Profiles

**NEW: Android Mobile Recorder**
• records phone calls, SMS
NEW: Hybrid JTAPI-SPAN recording

New recording mode using JTAPI for call signalling and SPAN for RTP traffic. This mode is especially useful when recording old phones or devices not enabled for forked recording.

NEW: Multi-Tenant capability

- highly secure, uses isolated schemas
- Per Tenant Administrators
- SuperAdmin can edit users from any tenant
- Storage Quotas for each tenant

NEW: PKCS call recording encryption

- the OS and SW administrator is not able to listen to calls
- per tenant certificates
- Certificate Generation Wizard

NEW: HTTPS support

NEW: Multi-Language support

- currently English and Romanian are supported
- customers can create their own translation files
- RTL support

Refactored Replication:

renamed to HQ-Branch Replication multi-level replication (same server can be both sender and receiver)
multi-tenant aware now uses HTTP/S transport

More changes

- Call Player now embeddable in IFRAME for easy integration
- Auto-Refresh for View Calls
- Installer and Updater: Support for Windows UAC
- Live Monitoring now works with G.729, G.722 and G.711A, (web and phone service)
- Reset Admin Password utility, including disable of LDAP auth
- Updated: Quick Search for Users page
- Updated: Software Update using CDN, SW update for slaves, out-of-process
- Updated all embedded components to latest stable versions (Java 7, Postgres 9.0. Flash 11.3)
- Manuals converted to HTML, edited with web CMS, white-labeled
- all the fixes present in 6
Version 8.0

8.0.20 (4-Jul-2014)
- CHANGE replication - if not HA, do not search for a matching call to add a second storage

8.0.18 (3-Jul-2014)
- FIX too large result query in cisco phone list request
- FIX check for G729 license for playback
- FIX send logs incorrect date display
- FIX Localized active calls license status

8.0.17 (2-Jul-2014)
- FIX: may localization fixes throughout the application
- FIX PBX list double click was bringing Add dialog instead of Edit dialog
- FIX site timer leaks
- FIX database queue / replicator statistics sync issues
- FIX call dumper statistics
- CHANGE: Lowered View Calls double click time to prevent accidentally double clicking
- FIX timeout issues for statistics
- CHANGE background convertor - distinct traces for g729 vs normal, site shows now the sum of those two statistics
- FIX date filter for browser with different timezone than recorder
- FIX re-align channel timestamp with the other channel in RTPStack after ssrc change.
- FIX null callHandler in Forked Recording
- FIX forked - traces for terminal changed event errors
- CHANGE check version thread - if master is not responding, check version on call-replay.com
- FIX jtaip files stored in a different folder instead of temp
- FIX site - updated unlicensed call status
- CHANGE: Restrict temporary files expiration time to be at least 1h

8.0.16 (26-Jun-2014)
- FIX file renaming on close conversation for unlicensed calls
- FIX close unfinished conversations at startup - fix for more than 100 calls
- FIX view active calls - stop refresh timer when changing page
- FIX made background transparent for some image files
- FIX replicator - do not try to get replication config if upload is disabled
- CHANGE - Unset RestartOnFailure when stopping the service and re-setting it after restart

8.0.15 (25-Jun-2014)
- NEW: Min Volume Free Space is now configurable in Storage->Storage Volumes.
- NEW: added japanese language.
- CHANGE: CallHandler - refactored stop.
- FIX: Close all open conversations at startup - fix for non existing files.
- FIX: delete calls from site.
- FIX: close conversation for unlicensed calls.
- FIX: demand email notification.
- FIX: removing rtp streams when monitoring error occured.
- FIX: PreConfiguration - fix Shrink Database 'No' answer.
- FIX: AutomaticRestart and modified CallReplay Service to auto-restart on failure.
• FIX: ignore monitoring errors while recording - allow packets to be written to file.

8.0.14 (24-Jun-2014)
• FIX: site speex playback crash for short files.
• FIX: do not propagate monitoring exceptions to the caller, let it write to the file.
• FIX: restore multiuser.
• FIX: channel synchronizer - renew flush timer when any error occurred.
• FIX: restore issues: localization, delete only current backup folder, copy/move files combo is now persistent.

8.0.13 (23-Jun-2014)
• NEW: logo hiding at app startup on a different language (when buttons are wider).
• NEW: Web API - concatenate multiple input files into one output file.
• NEW: replication thrift api - create session with protocol version.
• NEW: replication thrift api - prevent using unauthenticated sessions.
• NEW: replication thrift api - recorder creation with version.
• CHANGE: Default deny permissions for new Role.
• CHANGE: background convertor - wake up after settings change.
• CHANGE: prevent deleting active calls.
• SPEED: optimized diskcleaner and replicator queue.
• SPEED: delete calls optimization.
• FIX: Force role selection when creating user.
• FIX: Adding User UI fixes + translations.
• FIX: audit on delete call.
• FIX: restart only modified PBXs.
• FIX: Keystore folder creation.
• FIX: Transcoder hash computation.
• FIX: reinsert in conversion queue when not needed for conversion.
• FIX: misleading error message on forked recording wizard.
• FIX: background convertor get chunk for conversion when original codec is not stored in db.
• FIX: reinsert in conversion queue - do not insert for replication already existing rows.
• FIX: Importing cisco certificate yields no message.
• FIX: site speex seeking while playback.
• FIX: disk cleaner for default tenant.
• FIX: converting older calls to new format.
• FIX: active calls from phone service.

8.0.12 (13-Jun-2014)
• NEW: Mitel - inject tone.
• NEW: RTP stack now works with different codecs on the left and right channels.
• NEW: enable/disable context menu items for calls depending on context.
• FIX: NullPointerException while opening Active Calls.
• FIX: Active Calls - fix display for null party info.
• FIX: Replay calls - fixed user rights.
• FIX: site - fill questionnaire.
• FIX: site - fix questionnaire playback.
• FIX: diskcleaner - fix for expiration by date.
• FIX: quality reports.

8.0.11 (12-Jun-2014)
• NEW: index by global id on calls table.
• NEW: checkbox for high availability - check for time sync only for HA.
• CHANGE: When adding a new user, initialize it with Agent Role.
• FIX: removed 'Downgrade' tab from Software Updater - cannot be used anymore.
• FIX: Phone Service monitoring issue.

8.0.10 (10-Jun-2014)
• FIX: site template to accomodate low resolutions.

8.0.9 (10-Jun-2014)
• NEW: log all rpc method calls on replicator client.
• FIX: Audit date filter.
• FIX: Force G729 Recompressssion was showing 'already running'.
• FIX: Phone service - fixed login.
• FIX: replication of calls with missing file.
• FIX: replicator - check for upload errors.
• FIX: PBX reloading.

8.0.8 (5-Jun-2014)
• CHANGE: Qodec licenses for g729 and g729b add up and count as the same channel pool.
• CHANGE: replicator - allow config request for disabled upload.
• SPEED: removed variable row height to improve performance.
• SPEED: reduced date to string function calls by 60%.
• FIX: hash code changed during background compression.
• FIX: MiTel encryption keys were not written properly.
• FIX: replicator concurrency.
• FIX: checking branch version.
• FIX: no scrollbars when compact view is enabled.
• FIX: workaround for thrift client connection leak.
• FIX: Phone service - fix for exit button.
• FIX: Jtapi and SIP synchronzation.
• FIX: Skip file renaming when closing unlicensed calls.

8.0.7 (3-Jun-2014)
• NEW: Storage volume - show error message for non-writable folder.
• NEW: Restore - added manual restore button that restores from path.
• NEW: thrift - global management / tracing of exceptions.
• NEW: New filtering system for browsing calls.
• NEW: replicator with upload sessions.
• CHANGE: replication refactoring - check for password in a single function.
• FIX: Phone name not being displayed.
• FIX: Storage volume - fixed sql for deleting storage volume.
• FIX: SRTP info not properly decoding inline message.
• FIX: Prevent RTP capturing on SPAN when the call is not SPAN-created.
• FIX: missing png on Tags.
• FIX: delete call after uploaded.

8.0.6 (29-May-2014)
• NEW: replication over thrift only; on android, too
• FIX: volume storage mapping for direct web access

8.0.5 (23-May-2014)
- NEW: G729 playback acquires license resource preferentially to de detriment of background compression
- FIX: conversion from .3gpp to .spx in transcoder.
- FIX: calls recompression

### 8.0.4 (21-May-2014)
- CHANGE: store SRTP info for sip inline messages.

### 8.0.3 (21-May-2014)
- FIX: support for ssrc change before channels synchronization on RTP playback.
- FIX: date validators.
- FIX: version detection on installer.
- FIX: automatic restarting recorder.

### 8.0.1 (20-May-2014)
- CHANGE: Don't ask for app dir and db dir in case of upgrade. Upgrade info in Memo Page. DB Backup just before upgrade.
- CHANGE: Add db upgrade with MessageBox before starting installer.
- FIX: Support for codec changed during call.

### 8.0.0 (20-May-2014)
- CHANGE: Removed 'use readable file names' checkbox.
- CHANGE: SIP signalling - do not stop / start call when receiving INVITE/ACK for the same pbx call id.
- FIX: Fixed 'storage not found' exception.

### New Features

NEW: File Integrity - validates recordings hashes for playback, replication, restore after backup

NEW: High Availability recording:
- record and store multiple versions of calls, one for each active recorder, then deduplicate them. This required to split Calls table in Call and CallStorage.
- maintain a list of online replicating recorders and dispatch playback to online recorder
- player / call details: combobox to choose the version of audio to be played (when there are multiple versions)

NEW: Custom Renaming of File Recordings using a powerful template system.

NEW: Multi-HQ replication: sends the recording to multiple master servers.

NEW: Backup + Restore locally or remotely on Windows shares; Backup History and list of backups for each call

NEW: Active recording + Playback for IPTrade Turrets

NEW: Record and Monitor Cisco encrypted calls (SRTP) using Forked Recording and JTAPI signalling ( +monitoring )

### Optimizations

Database optimizations:
• eliminated most file names from the DB, they are now computed on-demand.
• moved background compression / replication errors to their corresponding queues
• compress some fields
• removed some unused fields
• end time -> duration

IOPS Optimizations
• one-step transcoding process, without intermediate files
• compute file hash while recording
• removed .call files from storage
• buffered rewriting of wav header
• permanent transcoding process (not reloading it for every file and intermediate file)
• .png file generated in memory while transcoding for playback from .rtp file
• .png and audio files are generated in the same request from browser
• transcoding in separate process - via thrift

Integration API
• Thrift over HTTP
• Transcoding API over HTTP
• JSON list of matching calls

Fixes
• FIX: remote playback crossdomain issue
• FIX: access password for audio files delivered directly from storage
• FIX: create white .png file instead of empty file for zero-length audio tracks.
• FIX: fix call stopping because of incorrect silence detection
• FIX: removed extra configuration from PBX protocols
• FIX: added call locking for parallel processing
• NEW: A-Law Voice Activity Detection

Third Party Libraries
• Upgrade to PostgreSQL 9.3
• Upgrade to VCRedist 2013
• Upgrade to Thrift 0.9.1
• Upgrade to MyBatis 3.2.3
Interfacing with CallReplay

Direct Database Access

CallReplay uses an embedded Postgres 9.0 database. To connect to it you can use PgAdmin III (Start / Programs / CallReplay / Support / PgAdmin III). It is also accesible from Java, C# and all programming languages which can access Postgres.

Port: Set during setup process, default 5432

User name: callreplay

Password: set during setup process, default CallReplay_2005

Automatic Login URL

http://SERVER:PORT/CallRecorder/?user=USER&password=PASSWORD&tenant=TENANT&lang=LANGUAGE

where SERVER is the call recorder ip/address, PORT is the configured server port (see CallReplay Site Configuration), TENANT is the registered name of the tenant (if not provided, the "default" tenant will be considered), USER is the login name and PASSWORD is the password in clear text.

If you do not want to provide clear password, you may discard &password=, in that case CallReplay will ask you for password and use provided TENANT and USER.

Also, you can use the base URL with POST method, using the same keys and values as in normal URL:

http://SERVER:PORT/CallRecorder/Login

Call Details

http://SERVER:PORT/CallRecorder/?callDetails=CALL_UUID&user=USER&password=PASSWORD&tenant=TENANT

where call_uuid is the uuid of the call, found in Advanced tab of Call Details.

Example:
Sometimes you may want to show only the Call Details dialog, in this case use the following url (note ‘nomenu’):

```
http://<server:port>/CallRecorder/?callDetails=<call uuid>&user=<user_name>&password=<clear_password>&tenant=<tenant_name>&nomenu
```

Observation: this is best used inside of an IFRAME

Example:
Call Playback

http://SERVER:PORT/CallRecorder/?player=CALL_UUID&user=USER&password=PASSWORD&tenant=TENANT

Observation: this is best used inside of an IFRAME
File Storage URL

http://HOST:PORT/CallRecorder/Storage/STORAGE_VOLUME_ID/CALL_PATH

where STORAGE_VOLUME_ID is the volume index number in the table storage_volumes and CALL_PATH is the path to the required file relative to storage volume, which can be found in the Call detail dialog.

Example: http://localhost:8080/CallRecorder/Storage/1/2012/06/20/2012-06-20_22-07-47_81267_8666685394-1.rtp

The storage web URL is protected by a password which is random by default but can be configured from the Storage > Advanced > Player Access Password. The user name is "player".

Transcoding URL

Searches the call in the database, then transcode it to the desired output format, and send a redirect response to the temporary URL. If there is more than one call matching the query, the first one is used.

Parameters below are optional, that is searches can be made using any subset of parameters, so there is no longer any need to connect directly to the CallReplay database:

http://host:port/CallRecorder/Transcoder?callId=GUID&wireCallId=x
&beginTime=yyyy.mm.dd-hh.mm.ss&endTime=yyyy.mm.dd-hh.mm.ss
&callerIP=x&callerPort=x&callerMAC=x&callerNumber=x&callerName=x
&calledIP=x&calledPort=x&calledMAC=x&calledNumber=x&calledName=x
&format=x

Parameters:

- Format:
  - spx or wav - redirects to audio files; when multiple callIds were requested, their audio files are concatenated and the response redirects to the resulted file
  - png - redirects to a .png image of audio file
  - json - returns a list of details for all calls matching the query, in json format
  - beginTime and endTime - calls will be searched for start time between these values (if provided).
  - pbxCallId is the SKINNY call ID or SIP call ID
  - callId is the database call GUID; multiple callIds may be requested
  - numbers and names are searched for text containing provided values
  - IPs, ports and MACs are searched for exact match

External Replication API

Downloads call details using HTTP requests. To use this feature, you must add a replication branch and configure it for Replication API.

http://HOST:PORT/CallRecorder/Replicator?lang=x&tenant=x&replicationQueueName=x&password=x&lastReplicationId=x&deletePrevious=<true/false>

Parameters:

- tenant - the tenant id from which to download call details; can be omitted for default tenant
- replicationQueueName - must contain the exact branch name (case sensitive)
- lastReplicationId - the last confirmed replication id - first time should be 0, the subsequent invocations should
contain the last received replication id in order to be deleted from the database and advance to next operation.
• deletePrevious - whether to delete the row with lastReplicationId or not
• password - connection password set in the configured replication branch
• lang - one of the 2 char language identifier supported by CallReplay (see login dialog in administration site); used to display error messages; can be omitted for english

Response for new call:

http://HOST:PORT/CallRecorder/Replicator?...&lastReplicationId=5
<CallRecording>
< ReplicationId>22</ReplicationId>
< Action>Create</Action>
< CdrId>20006863629</CdrId>
< CallRecordingId>eeef1fc2-3e33-1bd9-c247-f19f4b2ac8f</CallRecordingId>
< StartTime>2014-08-224 15:01:36.211</StartTime>
< DurationMS>13000</DurationMS>
</CallRecording>

Response for deleted call:

http://HOST:PORT/CallRecorder/Replicator?...&lastReplicationId=71
<CallRecording>
< ReplicationId>72</ReplicationId>
< Action>Delete</Action>
< CallRecordingId>ed29894d-4a34-2944-b827-a76e00bd6b6a</CallRecordingId>
</CallRecording>

Response when there are no more operations at this moment:

http://HOST:PORT/CallRecorder/Replicator?...&lastReplicationId=654
<CallRecording>
< Action>Null</Action>
< ReplicationId>654</ReplicationId>
</CallRecording>

In case of an error, status code 500 will be returned along with an error message (in the chosen language).