Cost Effective Deployment of VoIP Recording

Purpose

This white paper discusses and explains recording of Voice over IP (VoIP) telephony traffic. How can a company deploy VoIP recording with ease and at an affordable price?

Audience

Company's that host VoIP and would like to offer their clients a cost effective and easy to deploy VoIP voice recorder.

Why Record?

Traditionally recording has been deployed for three reasons:

1. Compliance – the law requires that the calls are recorded and stored
2. Risk – due to the nature of the call, recordings must be retained and in the event of litigation be stored.
3. Performance – Enterprises want to know how their agents are communicating with customers, and through these recording make improvements to their agent's performance.

Executive Summary

Voice over IP (VoIP) will become the dominant means of transmitting telephony conversations in the coming years. This move from traditional circuit-switched to packet-switched technology will have a significant impact on the systems used to record communications.

The key points surrounding a VoIP recording solution are summarized below:

1. Recording **must** be considered at the design and configuration stage of a VoIP network to ensure that packets can be routed to the recording nodes within the network.

2. VoIP recording requires **less hardware** and much **less cabling** than circuit switched Recording.

3. VoIP will allow recorders to be deployed on **industry standard hardware.**

4. Low capacity recorders will be sold as **software only solutions** whilst higher capacity units will use add-in cards to provide additional compression capability where needed.

5. VoIP will facilitate the provision of recording **services 'within the cloud'** which can be offered by the network providers and retrieval of calls will be accessed via browser technology.

6. VoIP recording will become just one aspect of **overall IP recording** solutions.

Background

Voice over IP is a means of transmitting speech over a packet-based infrastructure such as the Internet. The continuous stream of bytes representing the audio signals are chopped into (fairly) small packets – each holding, typically, twenty to thirty milliseconds of audio and sent across the data network to their destination. By sending a stream of such packets in each direction, two-way telephony can be achieved.

The technical issues around quality of transmission are not discussed here but obviously the compression algorithms used, the time taken to deliver the packets and the proportion that get through to the far end determine the overall quality of the 'call'. Given the ready availability of bandwidth, the inefficiencies of using IP packets to transmit relatively small payloads can now be largely ignored as the savings in other areas outweigh this.

Recording of VoIP

Having set the scene of VoIP's gradual introduction, how does this affect recording systems such as OrecX TR.

VoIP Recording Approaches

With a packet-switched network, the speech packets are transmitted over a path determined by the routers and switches in the network. In a complex network with multiple paths, it is difficult to predict where packets will be sent. Successive packets on a given connection can even follow different routes or arrive out of sequence.

Two basic approaches can be taken to recording calls in an IP environment:

1. "passive" – in which the recorder attempts to 'sniff' or 'snoop' the network without any of the parties involved in the call needing to be aware of this.

2. "active" – in which at least one of the parties involved in the call is ensuring that the required data is deliberately sent to the recorder

Impact of VoIP on Recording Systems

Hardware commodization

Since the voice packets are typically carried over standard fast Ethernet connections, there will no longer be a need for specialist telephony connection hardware. Hence we will see a gradual move to the use of standard PC chassis as recorders rather than the proprietary chassis of today. This should give greater flexibility and a lower hardware cost than today's solutions.

Distributed Call Centers

One of the uses of the above Application Service Provider (ASP) model is in building less formal and more distributed call centers. Since there is no longer a requirement for expensive ACD, IVR and other systems such as recorders, it becomes much more cost effective to deploy small teams of people at multiple sites whenever they are needed.

For example, back office staff in a branch may be used as an 'overflow' resource to assist the main call center when it is overloaded. In such environments, recording nodes can be placed in the remote offices and managed over the WAN or at the central office if all VoIP packets are routed past them there.

Home Working
The ultimate distributed call center is where individual users are working from home with nothing more than a data link into their company's network. Rather than having to have separate data and speech lines, VoIP can be used to allow operation over a single – albeit higher speed – line such as DSL or cable modem. In this environment, a 'personal' recorder application may be running on the CSR's PC and downloading recordings out of hours if bandwidth is limited. Alternatively, a shared recording node may be located at the central facility from which calls are routed out to CSRs' homes.

VoIP Recording Challenges
As with all new technologies, there are some technical challenges to overcome:

Routing
The most difficult aspect of recording in a VoIP infrastructure is in ensuring the packets reach a recorder. In some topologies, it will be difficult for recorders to reach all of the data. This has been discussed earlier and is largely dependent on the physical topology of the network and the sophistication of the routers and switches employed.

Bandwidth
Paradoxically, the move to VoIP both improves *and* exacerbates the bandwidth requirements of voice recording. In the short term, we expect to see much of the VoIP transmission being uncompressed as calls are only routed over the final, LAN segment on IP. In this case there is no shortage of bandwidth in the LAN and a desire not to lose any quality of speech – hence the normal toll quality 64kbps encoding will be retained. In these cases the recorder will need to compress the audio for efficient storage exactly as happens today in circuit switched systems. High capacity recorders will therefore require DSP cards to provide multi-channel compression.
In the medium term, more VoIP calls will originate in the internet or intranet and be carried as IP over long distances and often over limited bandwidth links to/from peoples homes. Here there is a real incentive for the voice to be compressed for transmission. In such cases, the recorder will typically store the recording in the compression format that it was transmitted in. Any minor savings to be made by decompressing it and recompressing it in a different standard will be outweighed by the resultant drop in quality. In the longer term, the ready availability of bandwidth is likely to drive people to higher quality transmission using higher sampling rates. Preserving this higher quality yet compressing the signal enough to give efficient transmission/storage will require different compression techniques.

Encryption
One of the biggest challenges to recording systems is the increasing use of encryption to secure transmissions from prying eyes. When encryption is used, it will become virtually impossible to build commercial recording systems without them linking in to the encryption engines so that the necessary decryption keys are stored along with the recordings. Where encrypted 'tunneling' is used, however, this is transparent to the

applications at each end and hence should not impact recording as long as the recorders access the packet data on the unencrypted ends of the transmission.

Screen Capture
Over time, it will become the norm to record the activity on the CSR's desktop along with the voice calls that they took. In a VoIP environment this becomes simpler and cleaner than at present. Note that the bandwidth required for accurate screen capture may well exceed that required for the speech recordings.

OrecX Solutions

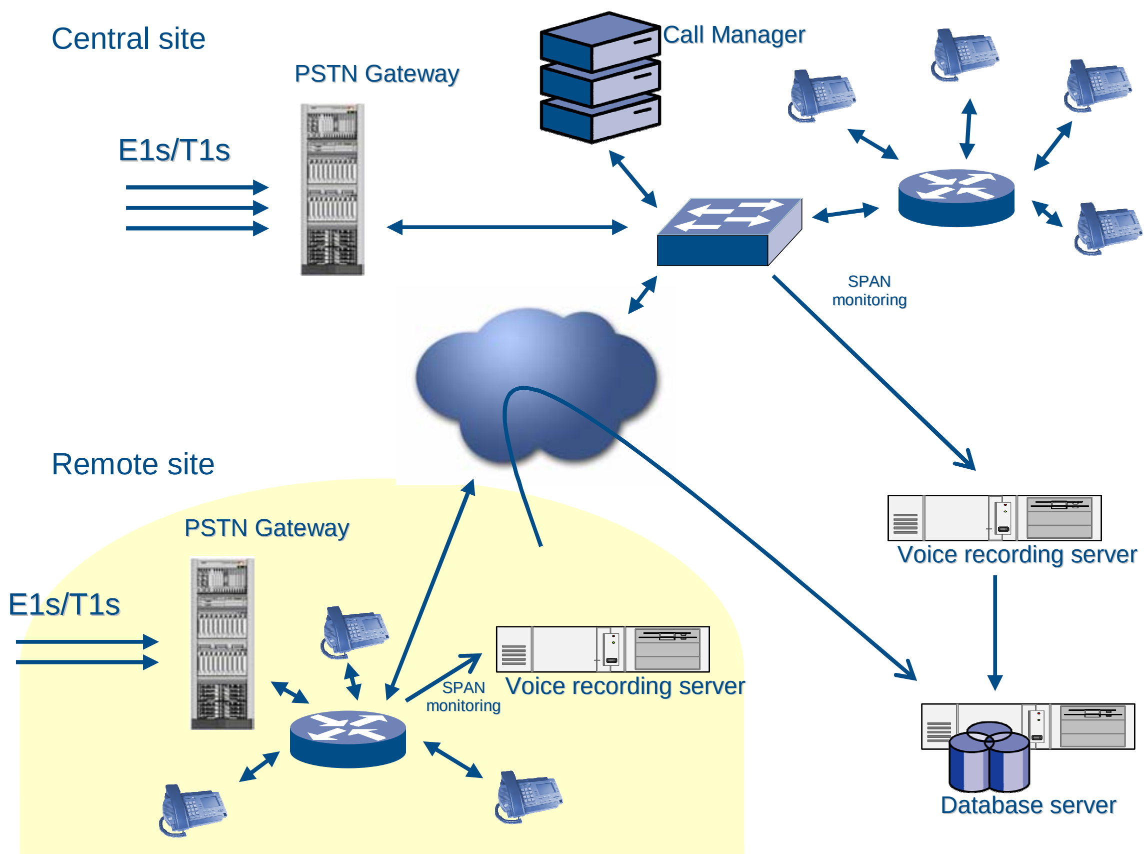
OrecX TR is the offspring of Oreka an open source VoIP call recording project posted on SourceForge.net. Customers wanted an easy to install – easy to use – browser based VoIP recorder. With over 10,000 users and 135,000 project web hits from developers around the world, OrecX is poised to make a significant impact on the voice recording world.

Why?

Browser Based Technology: recordings can be retrieved from anywhere at anytime with the right administration rights.

Fine grained privilege access system: Make sure employees in your organization access what is necessary and only what is necessary to carry out their daily tasks. Fine granularity with sensible defaults: Out of the box, the software sets privileges the way most organizations do. However, if you have any special needs, it is possible to attach users to any number of security profiles that define access level to each business objective. Complex organizational structures can easily be mapped to OrecX TR by using the hierarchical group system.

<u>Scalable – from 10 to 10,000 users single or multi-site</u>: There is no limit on the number of recording services reporting to the central database logger. The capturing process can happen anywhere between fully centralized to fully distributed as pictured on the system architecture diagram below.



System architecture diagram

<u>Low impact pilots</u>: no implementation costs or specialized hardware or software required. Pilot can be up and running within two hours.

<u>Low purchase price</u>: pay for only what you use on a monthly, annual or per call basis.

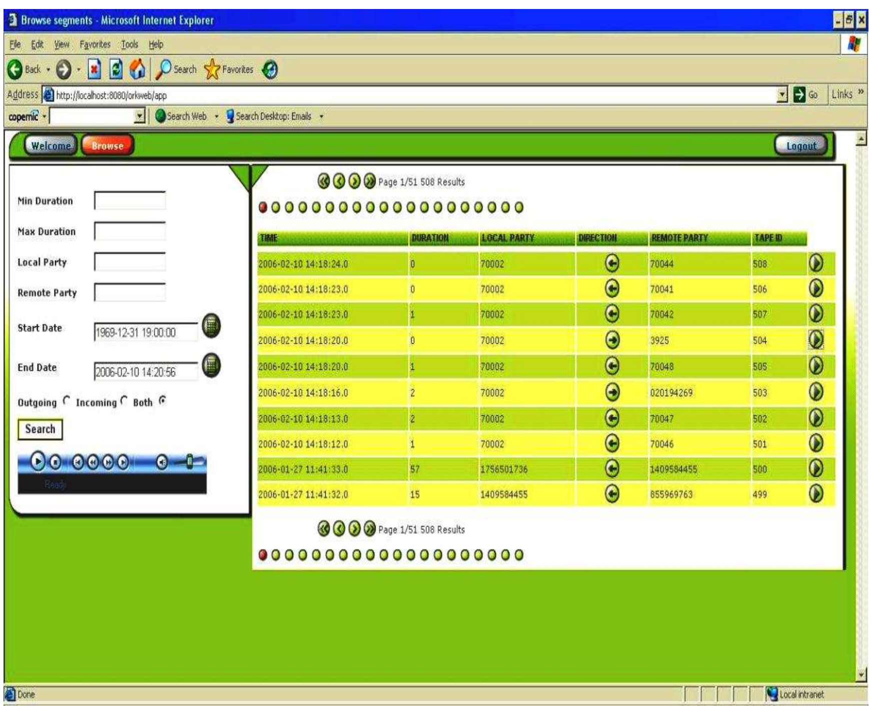
<u>Low implementation costs</u>: installs in hours with no specialists required.

Flexible tagging system: Attach any kind of information to your recordings manually or programmatically via the OrecX API, bookmark interesting places in calls and be able to search through all this personalized information instantly.

Open source, open API - Possible to develop your own extensions to the system - integrates well in existing CRM systems The web service API is easily usable from any programming language, without any special compiler or environment. It is as simple as requesting URLs and parsing XML in return. The Open source Plug in architecture allows organizations to add their own processing modules to OrecX.

Operating system agnostic OrecX runs on Windows and various flavors of Linux. Other posix operating systems can be supported on request.

Analytics ready - simple integration with other dbases to build detailed file names OrecX is compatible with all industry standard databases (Oracle and IBM DB2, MS-SQL...) and open source (PostgreSQL, MySQL). It is possible to have OrecX log to an existing database for integration into another business system in production.



About OrecX

OrecX is the primary developer and sponsor of the open source call recording initiative managed by Oreka at the oreka.org website and through the project posted on sourceforge.net.