Ultra ELECTRONICS | AudioSoft

White Paper

Solutions to VoIP (Voice over IP) Recording Deployment

A three point plan for updating your communications

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1.0 Executive Summary

Although Voice over IP (VoIP) has been around for several years, the benefits of integrated voice and data solutions and the prospect of lower cost, centralised equipment for business is one just being realised.

With this new method of voice communications follow new requirements to record this information.

This white paper describes the benefits and solutions to recording VoIP communications.

This document will:
- Describe VoIP and its benefits
- Outline solutions to recording VoIP
- Describe the procedure to be followed to successfully record your VoIP communications.

Audience

This white paper is relevant to:
- Those responsible for the operation of any recording system and who wish to know more about VoIP and VoIP recording
- Those whose networks either currently use VoIP or may do so in the future.

2.0 Step 1: Understand why VoIP is so popular

2.1 History

Voice over IP (Internet Protocol) began in 1995, allowing people to have voice conversations when only PC-to-PC communication was available by compressing voice data, translating it into packets and sending these packets over the internet.

In 1998, VoIP traffic was making up less than 1% of all voice traffic. In 2007, half of all calls (by international minutes) are expected to be VoIP:

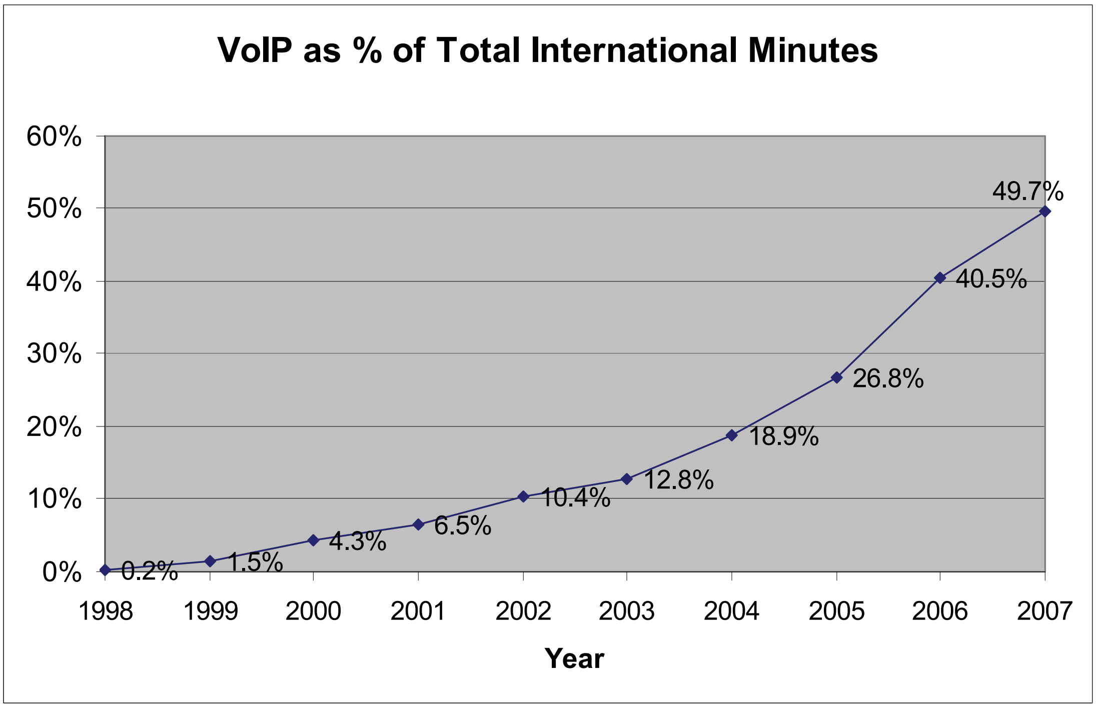


Figure 1: VoIP as % of Total International Minutes from 'recent analyst estimates of the growth of VoIP in terms of transmission' [1]

2.2 Sending Data in Packets

IP is a packet-based protocol; the data traffic is broken into small chunks which are sent individually to their destination. These packets can take different routes to their destination and therefore arrive at different times. A term used synonymously with 'IP' is 'TCP' which spawns the commonly used phrase 'TCP/IP'. TCP allows for lost packets to be resent and assembles them in the correct order at their destination, ensuring all information is received.

For voice conversations, it is not critical that every millisecond of speech is heard although it is important that it is heard in the right order. VoIP uses Real-Time Transport Protocol (RTP) to assemble the packets into the correct order and replace missing packets with others it has received.

For more information on VoIP, see [2], [3] or [4].

2.3 Different VoIP Protocols

Various VoIP protocols are used to send the voice data, including each of the following:

MGCP

Media Gateway Control Protocol (MGCP) is a master/slave protocol, where the gateways are expected to execute commands sent by the Call Agents. For more information, see [4].

H.323

The H.323 protocol provides a foundation for audio, video, and data communications across IP-based networks, including the Internet. For more information, see [4].

SIP

Session Initiation Protocol (SIP) is a simple application layer signalling protocol for VoIP implementations. SIP is designed to be independent of the lower-layer transport protocol and can be extended with additional capabilities and services. For more information, see [4].

Cisco Skinny Client Control Protocol (SCCP)

Skinny Client Control Protocol (SCCP) is a Cisco proprietary protocol used between Cisco Call Manager and Cisco VOIP phones. For more information, see [5].

Choice of Protocols

States [2], "initially H.323 was the most popular protocol, though in the 'local loop' it has since been surpassed by SIP. This was primarily due to the latter's better traversal of NAT and firewalls, although recent changes introduced for H.323 have removed this advantage". [6] describes H.323 as a "legacy signalling protocol", focussing on "modern protocols or industry standards… SIP and Cisco Skinny". With SIP and Cisco Skinny appearing to be usurping H.323 [7] and other protocols as the two competing industry leading standards, Cisco have now moved to using SIP for CCM 5.0 onwards.

2.4 So why use VoIP?

Using VoIP instead of traditional telephony can save costs in two major ways:

An internet connection is cheaper than a traditional long distance phone call. Whether you view a webpage on the other side of the world or in the next room, you are paying the same amount to do so, whatever your connection type. Traditional telephony, however, will be substantially more expensive to call internationally than locally and therefore the first advantage of VoIP becomes apparent.

IP infrastructure is integral to the business; traditional telephony is not. Most modern businesses and organisations have internet connections that provide functionality such as e-mail, web access and file transfer. This infrastructure can also be used for VoIP, thus removing the need for often expensive telephone systems and, in particular, the maintenance and support that go with them.

3.0 Step 2: Options to Record VoIP

3.1 VoIP Recording

When a call is initiated, a packet containing the call setup information is sent from the originating IP phone to the call manager which resolves this number to an IP address. These details are then sent back to the originator phone and the destination phone, and the connection is established. A similar procedure is adopted when the call is ended. All voice data packets also pass via the switch. On the basis that we wish to record all calls, the requirement therefore is to capture all required packets of information with the minimum number of recorders necessary.

It should be noted that as well as the reduced costs involved in a VoIP infrastructure, VoIP recording also requires less hardware and cabling than circuit-switched recording and so the costs are also reduced.

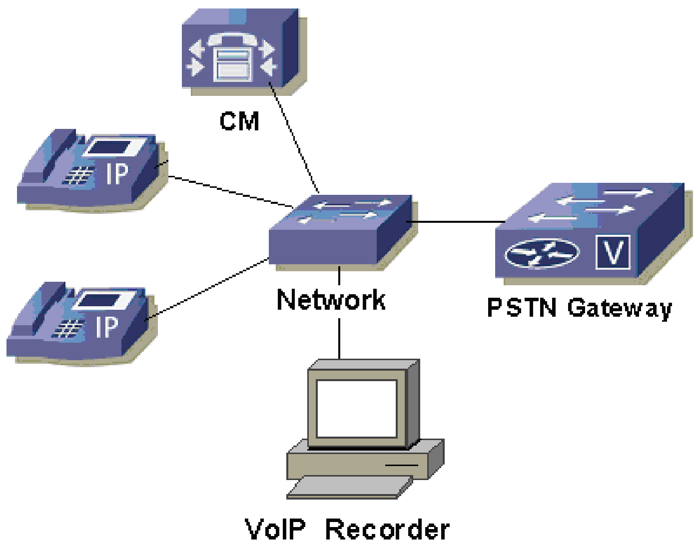
This section puts forward three different network topologies for VoIP recording and explains the advantages of each in different situations.

Option 1 – One site, centralised recording

Background

This configuration is suitable for a single site where one wishes to record all internal and external calls. All IP telephony components are distributed across one Local Area Network (LAN). Known as a cluster, this is the simplest of the configurations.

System Diagram



Features & Assumptions

- One recorder can extract all necessary information regarding who is calling who as well as the data itself
- All packets in this configuration must pass through one of the central switches, whether for call initiation, termination or voice data
- Each switch must be configured for spanning, which allows switches to daisy-chain through standard network infrastructure
- Option 1 is limited to a single site; options 2 and 3 are for multiple sites.

Replay

Replay via recorder through web interface

Cost Index

Cost index = 1 (where 1 = low, 5 = high)

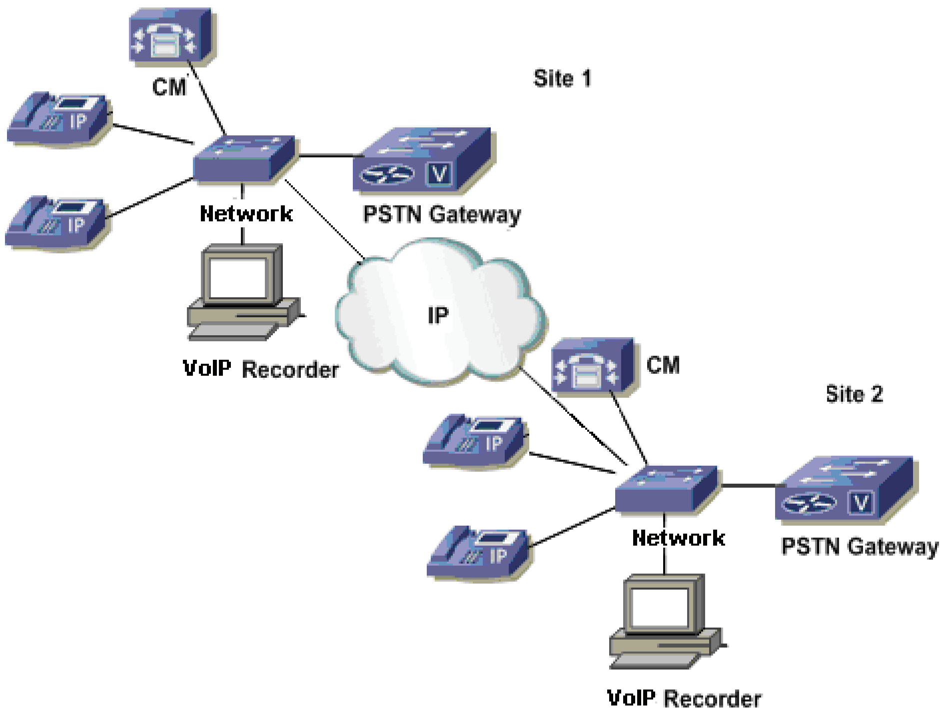
:

Option 2 – Multiple sites, distributed recording

Background

This may be a medium to large organisation with two or more offices in geographically separate locations, with each office connected via a WAN (Wide Area Network) and the multiple sites separated by IP (each site is a cluster). This configuration has IP telephony distributed across multiple LANs, connected via a WAN, with PSTN gateways and Call Managers (CM) on each LAN.

System Diagram



Features & Assumptions

- Suitable for geographically separated businesses
- All external calls coming in/out via the PSTN gateway will pass the local switch and therefore the local recorder
- For internal calls, all packets will pass both sets of switches and therefore be recorded twice; this can be resolved using database management
- Each switch must be configured for spanning as in configuration 1.
- Reduction in CCM infrastructure is also possible; this requires at least one PSTN gateway for each LAN and possibly one Call Manager controlling multiple sites

Replay

Replay via recorder or desktop PC

Cost Index

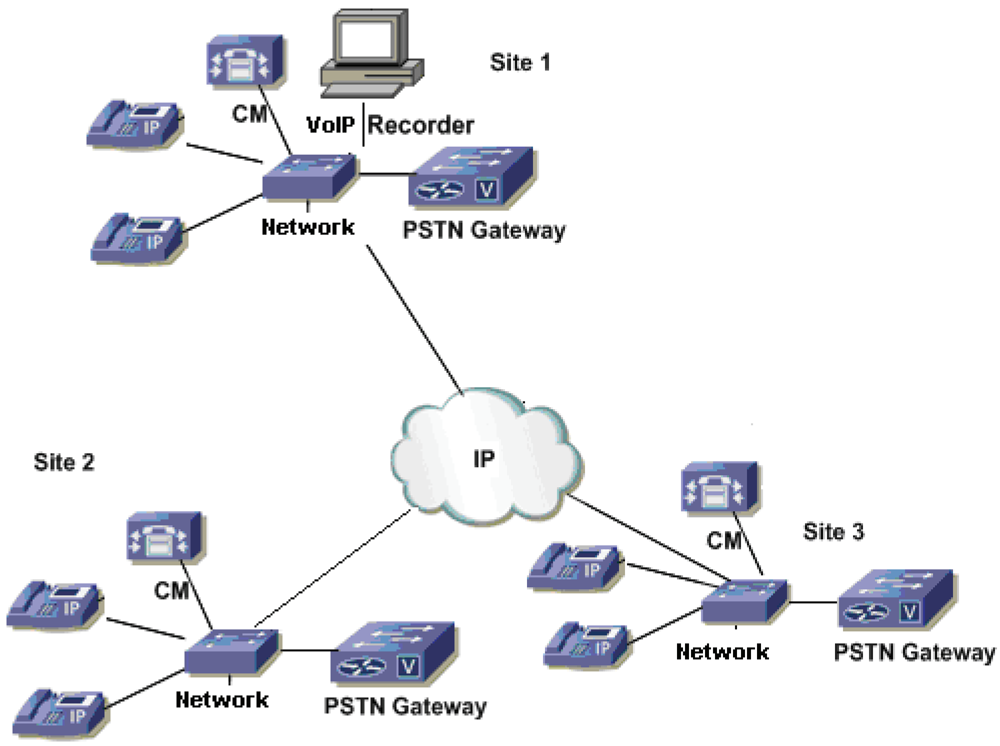
Cost index = 2 (where 1 = low, 5 = high)

Option 3 – Multiple sites, centralised recording

Background

Multiple sites, separated by IP; clusters may be distributed over multiple sites. This may be a medium or large organisation with multiple sites separated geographically, connected via a WAN IT data network configured for IP telephony. This configuration has IP telephony distributed across multiple LANs.

System Diagram



Features & Assumptions

- Suitable for a large organisation with many smaller sites as the number of recorders is minimised. Maintenance and support effort is also minimised.
- Centralised system allows easy monitoring, replay and management of remote sites.
- Existing network must fully support remote spanning (RSPAN) for robust implementation. Further network features are also required.
- Calls that are internal to any one site will not be recorded. These calls are not passed across the network so cannot be recorded from the central recording site.
- Network switches must be configured for remote spanning. Please consult your VoIP recording supplier for further information.
- Consideration must be given to how the data will be secured when transmitted across public networks.

Replay

Centralised replay via recorder or desktop PC

Cost Index

Cost index = 3 (where 1 = low, 5 = high)

3.6 Other Considerations

There are various other considerations when implementing a recording system on telephony networks of this nature.

Spanning

Spanning is the capability to copy data travelling down one port (or ports) to another port for the purpose of analyzing, and thus recording, the traffic.

Most Cisco switches are only configured for SPAN, meaning you can only collect packets that are seen by that switch, from that switch.

In order to record all calls, both external and internal, one needs a recorder on each of the network switches where there is an IP phone (or communications device). There are certain limitations imposed by these circumstances. The geographical separation between the relevant switches and the recorder is limited to 75m, this derives from the ethernet standard maximum cable length, although it is possible to achieve greater lengths using repeaters. However, if one is only recording external calls then one only needs a recording system at the switch by the Call Manager. In practice, 2-8 recording interfaces per recorder is typical, dependent upon the size of your requirements.

Effect on conference calls

Because VoIP recording can be done passively by ‘sniffing’ the network, there is minimal effect on network load. Rather than physically connecting to every switch, the network topology can be set up to use remote spanning to reduce the infrastructure required. In this topology, all packets are copied to the port on which the recorder is connected. This facilitates recording of all telephony calls on the system (both external and internal). “Soft calls” from one PC to another are not restricted in bandwidth in the same way that telephony calls (using standard IP handsets) are, to allow for video conferencing, file transfer and other PC-related interaction. However, because there is not this restriction in bandwidth then consideration needs to be given to how many ‘soft’ calls can be recorded without putting an unacceptable load on the network through passing too many calls of high bandwidth from one switch to another.

It should be noted that in VoIP recording, one generally records each side of the conversation separately, which is different from standard telephony recording in which the whole conversation is recorded in one file. The Call Manager does not need to exist on the local LAN as there is enough information within the protocol messages, for a call to the local LAN, to enable all voice streams associated with a call to be captured.

It should be noted that if the call is conferenced and the Call Manager is not on the local LAN, that you will only receive two voice streams, one for the local side and one for the conference side and you will not be able to disseminate the conference side into its individual voice streams. For example, suppose a remote user in the U.S. (without a Call Manager on the LAN), the Vice President of Sales, is conference calling the Sales team in the U.K. (each on separate handsets with a Call Manager on the LAN) then two streams of the conversation would be recorded instead of the traditional one from PSTN telephony recording or one per user that would be possible if there was a Call Manager on each LAN.

Encryption

If an IP telephony system is distributed across multiple LANs then you will need at least one recorder per LAN, and more if any LAN goes outside of the limitations.

If the VoIP network uses encryption then, due to its passive nature, it would not be possible to decrypt the data. Under some circumstances it may be possible to record the encrypted data with decryption carried out at the replay client rather than the recorder. If your network uses encryption then discuss this with your audio recording supplier.

Recommended Solution:

The recommended solution depends on your existing infrastructure, the size of your business and the need to record data other than VoIP.

	Configuration 1: One site, centralised recording	Configuration 2: Multiple sites, distributed recording	Configuration 3: Multiple sites, centralised recording
Effective for single sites	✓✓	✗	✗
Scaleable	✓	✓	✓✓
Simplicity of infrastructure	✓✓	✓✓	✓
All external calls recorded	✓✓	✓✓	✓✓
All internal calls recorded	✓✓	✓✓	✗
Ease of replay	✓✓	✓	✓✓
Cost Index	✓✓	✓	✓✓

Table 1: Analysis of different options for VoIP Recording

✗ = unsatisfactory, ✓ = satisfactory, ✓✓ = excellent

4.0 Step 3: What should I do?

- Detail your network topology and the equipment manufacturers used. This should include the make and model of switches and handsets, the signalling protocol used and details of any encryption as well as a diagram of the connections.
- Consider the network operation and usage and your specific requirements. In particular, do you require both external and internal calls to be recorded or just external calls?

If you have an existing VoIP telephony network and wish to record the calls:

- Consider how you would wish to replay the calls.
- Consult with your voice recording supplier as to how best to implement the recording solution with your existing infrastructure. Request a VoIP recording questionnaire in order to specify your requirements accurately.
- Depending on the size and the infrastructure, select with your voice recording supplier the most appropriate recording solution as described in Section 3.

If you are looking to upgrade your VoIP network and wish to record the calls:

- Discuss with your voice recording supplier how best to lay out the network infrastructure needed for configuration 1, 2 or 3 as described in Section 3. Request a VoIP recording questionnaire in order to specify your requirements accurately.
- Implement the selected recording configuration.

If you do not currently have a VoIP network:

- Perform a return-on-investment calculation to see how VoIP could potentially reduce your costs, both through call costs and through no longer having to maintain and support both a circuit-based telephony network and an IP network.
- Discuss with your voice recording supplier how best to integrate a VoIP network with a recording system for the size of your organisation. Depending upon the size of your organisation, select from configurations 1, 2 and 3.
- Request a VoIP recording questionnaire from your voice recording supplier in order to specify your requirements accurately.

5.0 Summary

Many businesses are switching to VoIP telephony systems to reduce costs and require VoIP call recording to back up this infrastructure.

Leading VoIP recording products can be scaled to provide the solution to either simple or complex VoIP infrastructures with the benefit of streaming to the desktop. They offer the flexibility of a single recording solution with the reliability of either distributed or centralised recorders.

A VoIP recording solution that addresses your specific requirements and is tailored to your network infrastructure should be chosen with your voice recording supplier. This will allow you to record the calls that you need and give the best return on investment.

6.0 References

[1] http://www.the-voip-network.com/voipmarket.html , accessed on 28/03/2007.
[2] http://en.wikipedia.org/wiki/Voice_over_IP, accessed on 02/04/2007.
[3] http://www.rogerdarlington.co.uk/VoIP.html, accessed on 02/04/2007.
[4] http://www.protocols.com/pbook/VoIPFamily.htm, accessed on 02/04/2007.
[5] http://www.javvin.com/protocolSCCP.html, accessed on 02/04/2007.
[6] Deri, Luca, "Open Source VoIP traffic monitoring", http://www.sane.nl/sane2006/program/final-papers/R3.pdf, accessed on 02/04/2007.
[7] http://www.cisco.com/univercd/home/home.htm, accessed on 03/04/2007.

7.0 Bibliography

CM – Call Manager
CCM – Call Centre Manager (Cisco term)
H323 – (an internet protocol)
IP – Internet Protocol
LAN – Local Area Network
MGCP – Media Gateway Control Protocol
PSTN – Public Switched Telephone Network
Remote Spanning – Remote spanning enables you to gather statistics from a switch that is remote to the analyzer. The packets are sent to the analyzer through a special VLAN.
Remote spanning works exactly like a regular span except that the data is sent across the wire to the analyzer on a special VLAN.
Remote Spanning RTP – Real-Time Transport Protocol
SCCP – (Cisco) Skinny Client Control Protocol
SIP – Session Initiation Protocol
Sniffing (or snooping) – the process by which a recorder passively taps the call
Spanning – the capability to copy data travelling down one port (or ports) to another port for the purpose of analyzing the traffic.
TCP – Transmission Control Protocol
TCP/IP – Transmission Control Protocol / Internet Protocol
UDP – User Datagram Protocol
WAN – Wide Area Network
VoIP – Voice over Internet Protocol
VLAN – Virtual LAN

