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White
Paper

Successfully Transitioning to VoIP Recording

Overview

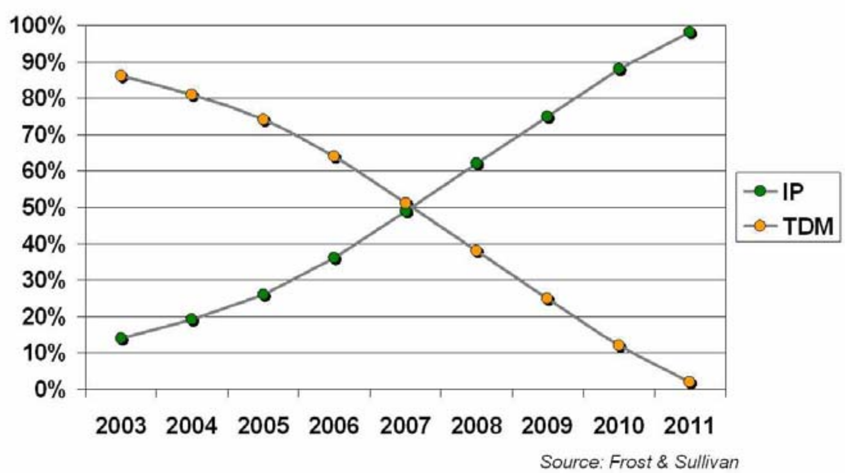
Contact center recording systems have become an essential component for managing the center's performance. As contact centers embrace VoIP it is crucial that the same high standards we have grown accustomed to when recording in a traditional telephony environment be maintained in the VoIP environment.

Fully understanding VoIP technology can be difficult given the many associated acronyms, buzz-words and differing vendor approaches. Having an overall understanding, however, is critically important as the impact of different architectures and designs can have a profound impact on recording system performance.

It is no longer a question of "if" the switch to IP telephony will occur in the contact center. It is now a matter of "when" – and it is likely to occur sooner rather than later.

A recent survey of 131 companies by Deloitte & Touche USA LLP found that two-thirds of businesses will be deploying VoIP technology by the end of 2006. In many cases, they're moving out of pilot projects and into full-fledged deployments.

A recent research study on North American IP Contact Centers conducted by Frost & Sullivan concludes that VoIP seats will equal the number of traditional TDM seats by 2007.



VoIP vs. TDM Contact Center Seats (North America)

This inevitable movement, driven by the cost and operational advantages that VoIP technology provides over traditional TDM telephony methods, necessitates sound planning to ensure that the operation of the center continues smoothly both during the transition and after.

Contact center recording systems have become an essential component for managing the performance of today's center. From managing agent performance through random monitoring to ensuring regulatory compliance by recording all calls, it is difficult to imagine running a modern contact center without these vital tools.

No matter whether the transition to VoIP occurs as a gradual migration, or a whole scale overhaul, it is critical to the continued success of the contact center operation that recording capabilities be maintained at the same level of functionality and reliability in the IP telephony environment as existed in the traditional telephony environment. To ensure a successful transition, it is important that contact center managers understand the key issues relating to recording in the VoIP environment.

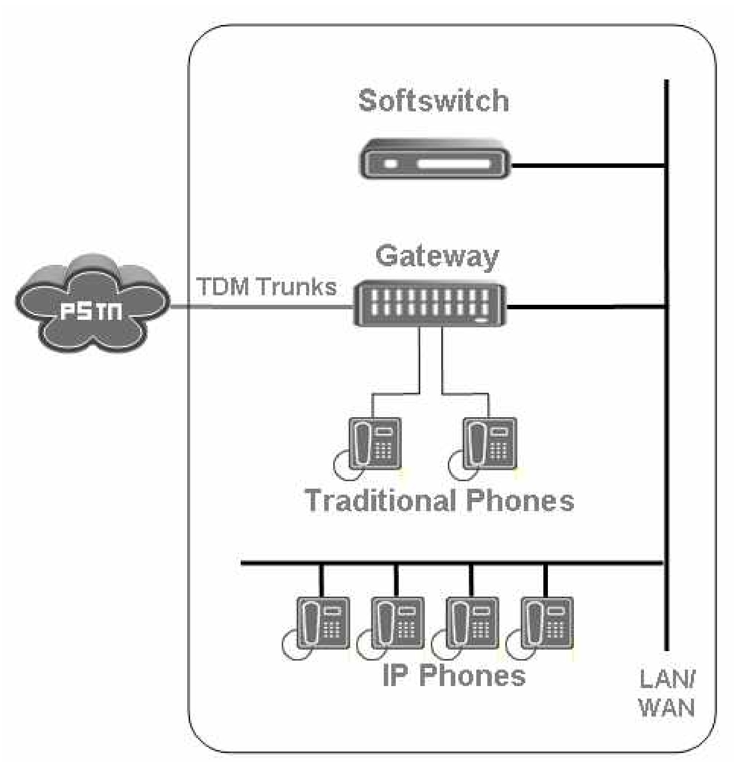
This white paper presents a look at the benefits that VoIP recording provides over traditional telephony approaches, different methods of recording in a VoIP environment and important issues to be considered during migration planning.

The Basics of VoIP

VoIP, or Voice over Internet Protocol, refers to a process of sending a voice conversation over the Internet, or more commonly a corporate LAN or WAN. In theory this is a fairly simple process. First the voice signal is split into separate data "packets". These packets are then transmitted over a data network to the desired destination. At the receiving end the separate packets are reassembled and the digital information is converted back into a duplicate of the original voice signal.

By contrast, in a traditional telephony environment a dedicated circuit is established between both ends of the conversation for the transmission of the voice signal. The disadvantage of having a dedicated circuit is that only one communication session at a time can use it. In a VoIP environment the total bandwidth of the network is shared by each session resulting in more effective resource usage.

Most VoIP implementations today are a hybrid mixture of traditional telephony and VoIP technologies. The vast majority of voice traffic, especially where the general public is concerned, travels over the traditional Public Switched Telephony Network (PSTN). Though this system is primarily based on digital transmission of the voice signal, connecting to a VoIP-based network requires the use of a VoIP gateway.



Typical VoIP System Configuration

The VoIP gateway serves as the bridge between the PSTN and the VoIP network. A gateway receives calls from the traditional network, generally through a T-1 or E-1 interface, converts them to VoIP data packets and routes them to the appropriate receiving device. The reverse occurs for conversation flowing in the opposite direction.

Another key component in a VoIP network is the Softswitch. The Softswitch coordinates call control functions, such as call setup, termination, routing and advanced features like conferencing. In some vendor implementations the Softswitch function is not positioned as a separate component, but contained within the Gateway itself. Regardless, the function of the Softswitch remains the same.

Although VoIP is simple in theory, in practice this process becomes much more complicated as issues such as network latency (the time delay between when a packet is sent and when it is received), jitter (packets arriving at the destination in a different sequence than they were sent), bandwidth (the amount of data that can be simultaneously sent over the network) and the potential for lost data packets can severely disrupt a VoIP transmission.

Almost all systems today utilize a specialized protocol, the Real-Time Transport Protocol or RTP, to help mitigate the impact of these and other physical network attributes. Having a common transmission protocol to deal with aids greatly in the recording of VoIP conversations, as will be seen later on.

H.323 vs. SIP

There are two other standards associated with VoIP, H.323 and SIP, that you will likely encounter when looking at this technology. In simplified terms, these protocols establish and control the VoIP connection between sending and receiving points.

H.323 was an early standard and is still very popular. SIP, which stands for Session Initiation Protocol, is a newer standard that is quickly gaining support. The merits of H.323 and SIP are often debated and they do compete with each other. Many industry professionals feel that SIP is the better approach, primarily due to its simplicity, flexibility and scalability. Many of the dominant vendors in the VoIP market also appear to be committed to SIP, including Cisco, 3Com, Nortel and Avaya.

A major advantage of SIP over H.323 is its ability to support new features considered to be an important component of emerging IP applications. H.323 was created to provide a comprehensive standard for voice and video transmission over an IP network. In addition to those functions, SIP also handles instant messaging, presence and e-mail.

SIP is still a developing protocol; however, and does have some shortcomings. It does not yet have the comprehensive set of call control features of H.323, in particular some needed to perform remote monitoring and control functions. There are also some issues concerning its ability to provide advanced security measures.

Vendors are aggressively working with the standards organizations to address these weaknesses, and it is likely just a matter of time until SIP emerges as the dominate standard for VoIP and other IP-based communications.

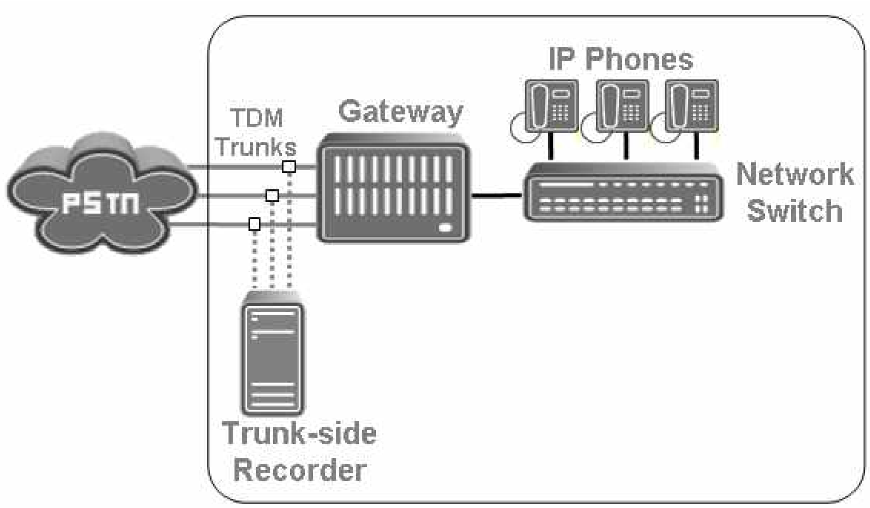
At the highest level VoIP and traditional telephony methods provide much the same functionality, which is delivering a voice communication channel between two points. However, the lower level differences that exist between the two approaches drive many of the business benefits associated with transitioning to a VoIP environment.

Comparing VoIP and Traditional Recording Methods

There are three basic approaches for recording in a traditional TDM telephony environment, direct tapping of the T-1/E-1 trunk lines (commonly referred to as “trunk side” recording), direct tapping of the extension lines (“extension side” recording) or using service observation or call conferencing capabilities in association with an ACD. Similar choices are available in the VoIP environment as will be discussed shortly.

In all of these methods information about individual calls is obtained through a call control service data link. This link is also used to establish an observation session when using that recording method. In a VoIP environment, this functionality is often provided by a service running on the Softswitch or Gateway.

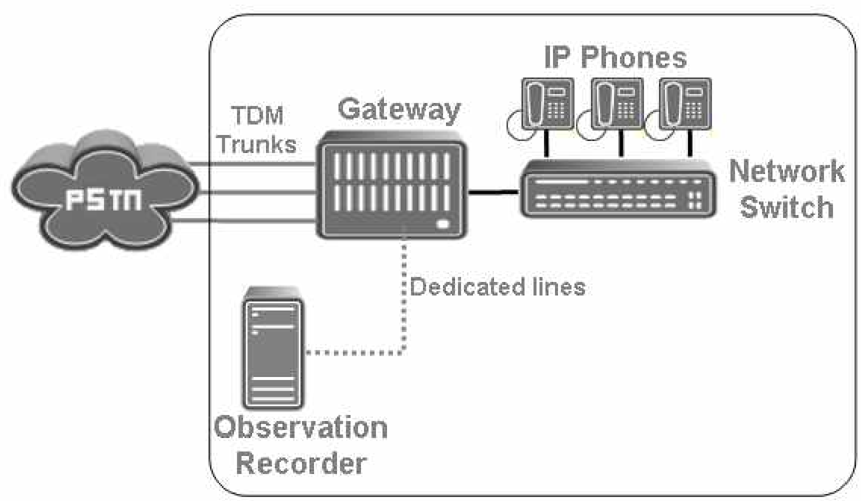
The first decision to be made when looking at ways to record VoIP calls is whether a VoIP recording system is even actually needed.



Trunk-side VoIP Recording using a Traditional Voice Recorder

If all of an operation's calls are carried on traditional PSTN lines, a traditional trunk-side recording method can be used to record all calls where they physically enter the center, including those occurring on VoIP phones. This approach does require that the VoIP gateway is capable of sending the needed call identification information to the recording system, rather it is obtained from the traditional CTI link.

Similarly, if the VoIP system can support observation or conferencing capabilities through non-VoIP lines, this too makes a traditional recording approach an option for recording VoIP calls.



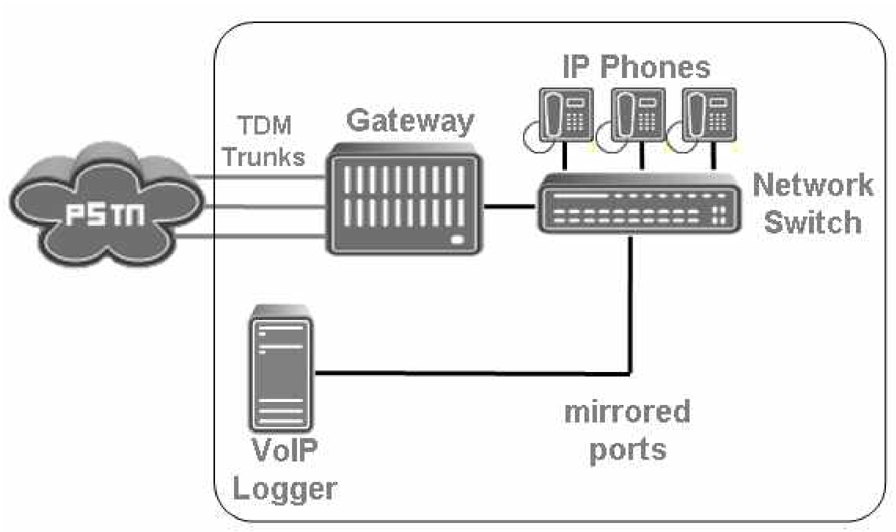
Observation VoIP Recording using a Traditional Voice Recorder

If traditional recording methods can be used to record calls in a VoIP environment why then would an enterprise choose to implement VoIP recording systems? Answering that question requires a look at how VoIP recording methods differ from traditional ones.

Passive vs. Active VoIP Recording Methods

Similar to the case with traditional recording methods there are different approaches to recording in a VoIP system. These are generally referred to as passive and active recording methods. The passive approach, also called "sniffing", utilizes a commonly found feature of IP network switches called port mirroring. This feature provides the capability to copy data packets from one port on the switch to another destination on the network. In a

Cisco environment this feature is called SPAN, which is short for Switched Port ANalyzer.



Passive Recording using Port Mirroring Capabilities

Although this feature was originally intended to provide monitoring and diagnostic support in a network, it is very useful as a way to record VoIP conversations. Using this method, the switches in a network are configured to mirror data packets from VoIP phone ports to ports on a VoIP recording system. The VoIP recording system decodes the incoming data packets to reassemble the original conversations and record them. This approach is feasible because almost all vendors use RTP as the protocol for sending VoIP packets.

In order to make sense of the packets being received, the recording system must have a way of mapping the VoIP data streams to the physical agents' stations or extensions. This is usually done using the IP address of the agents' VoIP phones. This works great in a "static" IP environment where the address assigned to a particular device in the network does not change. However, most business networks today utilize "dynamic" addressing where the actual IP address assigned to a device can potentially change at any time.

A passive recording approach in a dynamic IP environment is a possible solution; however, the IP address to extension mapping method must account for the potential changes in IP addresses. One alternative is to use the MAC address for the device. The MAC, or Media Access Control, address is associated with the network interface card. It is assigned during the card's manufacture and does not change. Determining and maintaining the MAC database for all VoIP devices in an organization can in itself be a significant effort, however.

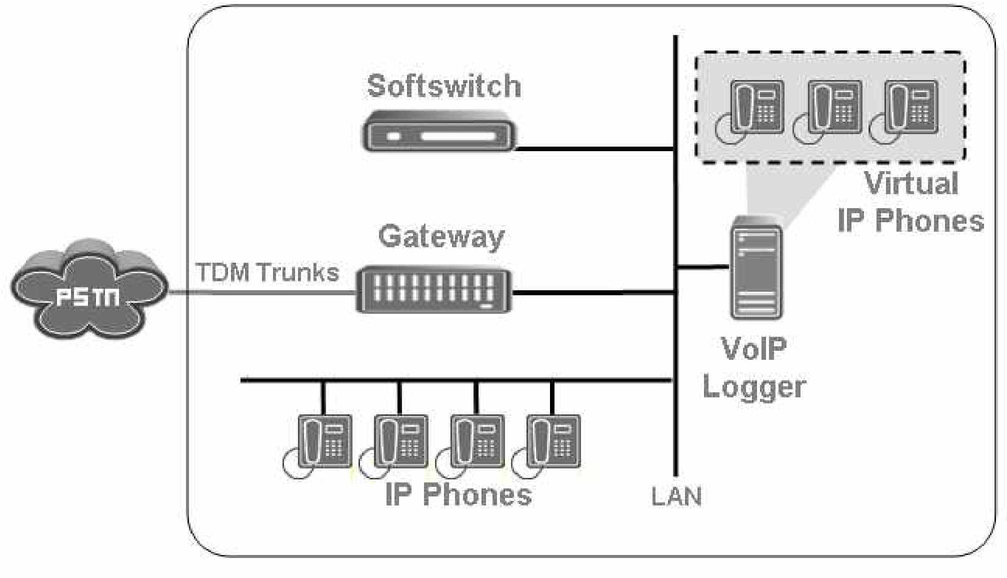
Another alternative for mapping devices in a passive recording environment is to detect and decode the VoIP call control protocol packets. This approach is subject to missing an occasional control packet and as a result, missing an entire call. The most dependable alternative for obtaining device mapping information in a passive approach is to get the information directly from the Softswitch or Call Control Service, assuming that a suitable application program interface, or API, is available.

One drawback to using the “sniffing” approach is the fact that the mirrored traffic is not filtered by the network switch. In other words, all data packets are forwarded to the recording device, not just the VoIP data packets. The recording device must go through all the packets being received, discarding those that are not related to VoIP. In environments with high levels of non-voice data traffic this can significantly impact the number of ports that can be handled by a single VoIP recorder, which ultimately adds to the cost of the system.

Increasingly, VoIP packets are encrypted within a network for data security and privacy reasons. This can be a potential problem when choosing to use the passive recording approach if there is no public key with which to decode the transmissions.

Another potential issue may arise due to limitations on a particular network switch’s port mirroring ability or capacity. IT departments may also be reluctant to endorse this particular approach due to the overhead of configuring and maintaining the port mirroring configurations on multiple switches in the network. Finally, in cases where network redundancy needs dictate the use of two or more data centers a call handled by the VoIP gateway in one center may not even be visible to a recorder connected to the gateway in the another center. This type of distributed carrier access usually makes a passive approach for VoIP recording unfeasible.

Using active recording in the VoIP environment resolves many of the limitations cited in using passive recording approach. An active recording method relies on the capabilities of the VoIP system to transmit duplicate data packets for calls to the recording device. Depending on the vendor, this may be accomplished either through monitoring/conferencing capabilities of the Softswitch or by packet forwarding capabilities of the VoIP phones or softphones.



Active Recording using Virtual IP Phones

In an active VoIP recording system the recorder interacts directly with the Softswitch to trigger the forwarding of VoIP packets directly to the recording device. Depending on the specific manufacturer's implementation the actual packets may be sent from the Softswitch itself or forwarded from the VoIP telephone device. In both cases the active recording process operates in substantially the same manner.

When the VoIP recording system is initialized it establishes a bank of virtual VoIP telephones. These virtual phones will be used as a resource pool to monitor or record the conversations. When a call is received, the gateway then causes a duplication of the VoIP data stream to be sent to one of the virtual phones in the recording device. With this approach device mapping is no longer an issues since it is maintained by the gateway.

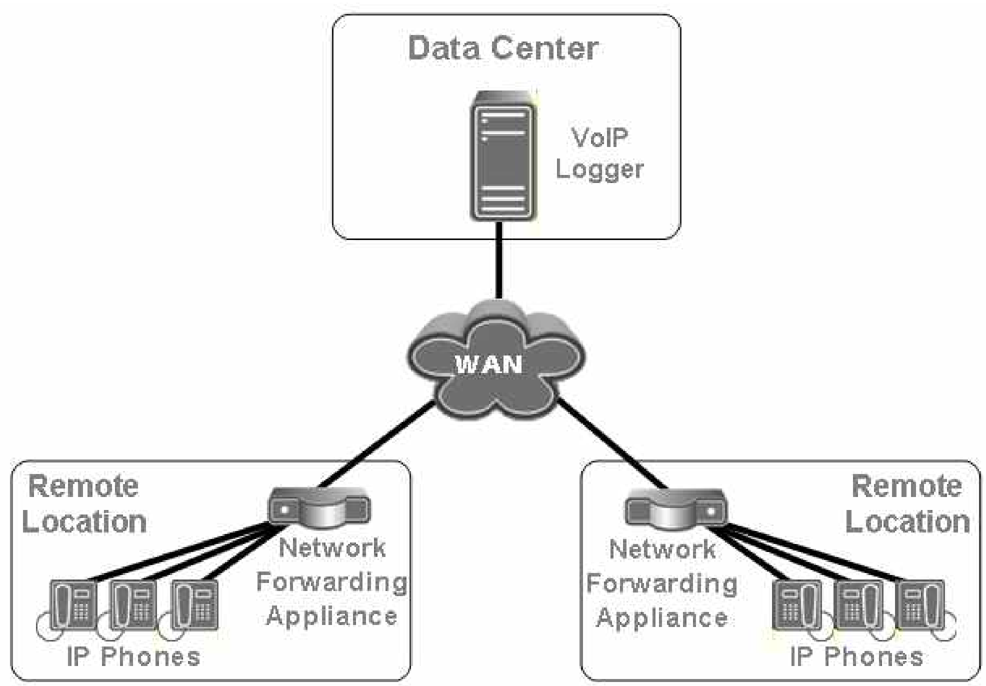
An active recording approach works as well in a distributed or multisite environment as in a single-site configuration. This allows for the centralization of recording resources, resulting in potential savings associated with implementing and maintaining the recording system.

With some vendor implementations the phones being monitored can be of any type: IP, digital or analog. This can result in additional advantages when recording in a mixed-type, or hybrid, telephony system.

A Third Approach – the Network Forwarding Appliance

A recently introduced approach to VoIP recording uses a Network Forwarding Device, or NFA, to provide a hybrid between the passive and active recording methods. An NFA is a high-reliability, remotely configurable network device that filters and captures VoIP traffic from the network and then forwards it to a VoIP recording device.

When deployed in remote branches the NFA provides a cost-effective way to capture and forward VoIP conversations without requiring packet mirroring to be implemented in network switches. The high reliability and remote configuration capabilities of this approach serve to greatly reduce the maintenance and support burden associated with implementing recording in a wide-spread business environment.



VoIP Recording using Network Forwarding Appliance

By filtering out and forwarding only the VoIP data packets, the NFA reduces the network bandwidth impact compared to standard mirroring approaches where all data packets are forwarded. When operating in an "active" mode, where local VoIP traffic is selected for forwarding based on information about the call, network traffic is reduced even further.

The NFA can also maintain the mapping of local IP addresses to physical stations, even in a dynamic addressing environment. This feature improves the reliability of a passive-style recording method in a distributed environment.

In a centralized data center the NFA provides advantages by being able to deal with the background network traffic that needs to be filtered before being recorded. This allows more effective use of more costly VoIP recorder resources.

Benefits of VoIP Recording

VoIP recording solutions provide specific advantages over traditional telephony recording approaches. These advantages often justify the costs associated with migrating to a full VoIP recording system even in cases where there exists a mixture of VoIP and traditional telephony devices.

Some of these advantages include:

- *Centralized Recording* – Removes the need to implement recording capabilities at remote sites and provides for more efficient use of recording resources. System maintenance and support requirements are also lower.
- *Faster Implementation* – There is no need to re-wiring or tap into traditional telephony wiring. Configuration is also typically much easier.
- *Reduced Maintenance Costs* – Moves, adds and changes can be implemented without the need for cable rewiring, punch downs or cross connects.
- *Scalability* – Adding additional recording channels can usually be done by simply expanding a software license. The number of recording channels supported per recorder is generally greater than for traditional telephony recorders.
- *Cost-Effective Reliability* – Redundancy can be achieved without requiring special hardware for audio switching.
- *Remote Branch and Home Agent Support* - The unique ability of VoIP telephony to easily support remote-branch or at-home agents greatly simplifies the process of extending recording capabilities to these locations.

Keys to a Successful Transition

Businesses follow a variety of strategies when deploying VoIP technology into their operation. In many cases VoIP is implemented when adding a new operational center or renovating an existing one. Although in some instances it is a total conversion, in most cases the transition will involve a possibly extended period where both VoIP and traditional telephony systems must operate side-by-side.

The information contained in this paper provides insight into some of the issues and tradeoffs that may be encountered during a VoIP implementation. No two projects are the same, however, as business requirements, operational needs and existing configurations can vary widely. The following general suggestions can also help in ensuring a smooth and successful transition to VoIP recording:

- Engage with IT as early as possible during the planning process to get advanced insight into potentially complicating issues. Work together with IT to decide which recording approach works best for your environment and business needs.
- Make sure you have a good business case for implementing a VoIP-based system. In some cases, a traditional voice recording system can accommodate the addition of VoIP telephones.
- Consider starting small if possible. Implement a single site to better understand the real-world issues you will be dealing with.
- Make certain the recording method aligns with your business needs. For example, don't rely solely on a trunk-side recording approach if you have a business need to record calls between extensions.
- Check to see that an accurate and complete network topology diagram is available indicating what levels of access are provided across different points of the network. In cases where "visibility" is restricted, multiple recorders may be needed to capture all calls.
- Confirm that network resiliency and redundancy issues are addressed to ensure reliability.
- Do due diligence in selecting a vendor. Check references, ideally ones with similar operational configurations to make sure the vendor is capable of doing what they represent. In hybrid (mixed IP and traditional telephony) approaches make sure the recording system can integrate both telephony types into a single environment.

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- Test, test, test… as much as possible. Try to assemble a fully functional lab environment, one that replicates as accurately as possible the planned one.

The NICE Solution

NICE pioneered and patented VoIP recording, and has the most comprehensive VoIP recording solution in the industry. The flexibility and scalability of the NICE solution enable it to easily adapt to the full range and scope of VoIP configurations – from smaller contact center operations to large, multi-site, multi-branch, high-end environments.

Certified by the world's leading VoIP switch vendors, including Alcatel, Avaya, Cisco, Nortel and Siemens, the NICE VoIP solution enables organizations to seamlessly transition compliance and quality management voice recording needs to the VoIP environment.

NICE is the acknowledged leader in providing IP-based recording solutions for contact centers. The NICE VoIP solution offers an impressive range of advantages including:

- A unified architecture for all analog, TDM and VoIP environments, even those incorporating systems from different vendors.
- Support for hybrid VoIP and traditional telephony environments in one system, enabling a smooth migration to VoIP. Call information and recordings are available during and after the migration, regardless of what telephony environment they originated from.
- A unified user interface, making the switch to VoIP totally transparent to the user.
- VoIP recording that is as reliable as traditional recording, making it extremely suitable for even the most demanding, mission-critical applications.
- NICE's Voice Recording Gateway, a network forwarding appliance, enables active VoIP recording in a passive environment and supports the recording of VoIP trunks.
- A software-only system installed on commercial-off-the-shelf (COTS) servers.
- Optional high-end redundancy providing increased system survivability and the elimination of single points of failure.

- The best performance in terms of recording channels per server, reducing the number of hardware components and increasing reliability.
- Proven scalability to thousands of recorded IP ports per site – NICE is the market leader in high-end VoIP installations.

NICE offers a true global presence and the industry's largest service organization, with partners and distributors in more than 100 countries serving over 23,000 customers. Visit www.nice.com to find out how the NICE VoIP recording solution can help your organization can realize the benefits of VoIP recording.

VoIP Glossary

Bandwidth	The amount of data that can be transmitted over a network for a given period of time, typically one second.
Circuit-switched Network	A communication system that establishes a dedicated channel for each transmission. The PSTN uses circuit-switching. Dedicated channels provide good reliability and quality, but the downside is that only one type of communication can use the channel at any given time.
Codec	An algorithm for encoding/decoding a signal into data packets. A codec also performs compression of the signal to reduce the amount of bandwidth required to transmit the signal. Generally the less bandwidth required the lower the quality of the resulting audio. The most popular codecs for VoIP are G.723.1 and G.729.
CTI Computer Telephony Integration	Originally intended as a way to combine computer and telephone applications. CTI is now more commonly associated with ACD systems and provides detailed information about calls such as the calling and called numbers and events such as holds and transfers.
E1	The European standard for a dedicated, point-to-point digital circuit. There are 30 separate channels in one E1 trunk. The primary difference between E1 and T1 circuits are the number of channels. The two standards are not compatible.
G.723.1	A codec standard for speech compression operates at 6.4 kbits/sec or 5.3 kbits/sec.
G.729-	A codec standard for speech compression operates at 8 kbits/sec. There are two compatible versions of G.729: G.729 and G.729 A.
H.323	A standard for real-time voice, video and data communication over packet-based networks such as the Internet. H.323 addresses problems inherent to packet-switched networks such as packet delay and loss. The SIP standard is another common protocol for VoIP communication.
Internet Telephony	A general term for technologies that send voice, fax and other forms of information over the Internet. Although commonly used interchangeably with VoIP and IP Telephony this is not technically correct.

IP PBX	IP PBX is a customer premises IP-based telephone system that manages telephones in the enterprise and acts as the gateway to external networks.
IP Phone	A telephone handset that connects directly to a PC or an IP PBX via a network. IP phones generally look and perform much like traditional telephones. Also known as handset.
IP Telephony (Internet Protocol Telephony)	A general term for the technologies that use the Internet Protocol's packet-switched connections to exchange voice, fax and other forms of information that have traditionally been carried over the PSTN. Although commonly used interchangeable with VoIP and Internet Telephony this is not technically correct.
IP (Internet Protocol)	A specification that defines the transmission of information over data networks. It tracks the Internet addresses of nodes, routes outgoing messages and recognizes incoming messages.
Jitter	An effect of network delay and routing that causes data packets to be received in a different order than which they were sent. A jitter-buffer is generally used on the receiving end to restore the packets to their correct order before processing.
Latency	Also called delay. The amount of time it takes a packet to travel from source to destination. Together, latency and bandwidth define the speed and capacity of a network.
MAC Media Access Control	A MAC address uniquely identifies the interface hardware, such as an Ethernet adapter, that connects a computing device to the network. It is usually assigned by the manufacturer when the device is made.
Network Appliance	A computing device operating on a network that is limited to a single or specific function. Network appliances are typically designed with redundancy and remote configuration capabilities to reduce support and maintenance requirements.
Packet Sniffing	Refers to the act of capturing data packets transmitted over a network. It usually implies a passive approach where the network is unaware, and unaffected by its presence.
Packet-switched	A communication system that breaks information into small packets before sending them. Each packet can follow its own path to the destination. Packet-switched networks are subject to latency and packet loss, but provide for more efficient use of bandwidth compared to circuit-switched networks.

POTS (Plain Old Telephone Service)	The term refers to the standard telephone service that most homes use. Also called the PSTN.
Presence	In a VoIP environment, presence refers to the ability to determine another person's availability for establishing a communication session. It commonly implies going beyond whether a person is simply on-line to what state or activity they are currently involved in.
PSTN (Public Switched Telephone Network)	The network of wires, signals and switches that lets one telephone connect to another anywhere in the world. Also called the Plain Old Telephone Service (POTS).
QoS (Quality of Service)	Measure of performance for a network to deliver traffic with minimum delay and maximum availability. Latency, packet loss, network jitter, and many other factors contribute to QoS measurements. In VoIP, QoS generally refers to the quality of the voice call. Newer network technologies address VoIP QoS by prioritizing the delivery of voice data packets over data packets.
RTP (Real-Time Transport Protocol)	A commonly used method for transmitting voice packets over an IP network. RTP provides services such as time stamping and delivery monitoring. Time stamping allows received packets to be assembled in the same order they were transmitted.
SIP (Session Initiation Protocol)	A protocol that provides telephony services similar to H.323, but is less complex and uses fewer resources. SIP uses text format messages to setup, manage and terminate multimedia communication sessions.
Softphone	A software-based IP phone application that runs on a PC or other computing device. The application provides a wide range of features typically found in business telephones. In addition it can integrate with other PC applications. Softphones are typically used with a headset and microphone connected to the computer.
Softswitch	Generally refers to the application that performs the call control functions in a VoIP system. Depending on a specific vendor's implementation, the softswitch may be contained within the VoIP Gateway or operate in a separate device. Call control relates to the setup and termination of calls, including call routing.

SPAN Switched Port ANalyzer	A Cisco specific implementation of network switch port mirroring. This feature was originally designed to support remote analysis and diagnosis of network transmission problems.
T1	The North American standard for a dedicated, point-to-point digital circuit. There are 24 separate channels in one T1 trunk. The primary difference between T1 and E1 circuits are the number of channels. The two standards are not compatible.
TDM (Time Division Multiplexing)	A data communication approach that divides the available bandwidth of a digital circuit into separate time slots to provide multiple channels. TDM is used by T1 and E1 circuits.
Trunk	A communications channel between two points. Each T1 or E1 circuit is typically referring to as a single trunk.
VoIP Gateway	A network device that converts PSTN traffic to VoIP traffic. The primary functions of a VoIP gateway include voice compression/ decompression, packetization and forwarding. A VoIP gateway operates in conjunction with a call control device, or softswitch, to establish and manage the connection to IP phones or softphones.
VoIP (Voice over Internet Protocol)	The capability to carry normal telephony-style voice over an IP-based data network. In VoIP, the voice signal is encoded into data packets and transported over the network using various standards including H.323 and SIP.

About NICE

NICE Systems (NASDAQ: NICE) is the leading provider of Insight from Interactions™, based on advanced content analytics of traditional and IP telephony, Web, radio and video communications. NICE solutions improve business and operational performance, as well as security. NICE has over 23,000 customers in 100 countries, including the world's top 10 banks and over 75% of the Fortune 100 companies. For more information visit the company's Website, www.nice.com.

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