White Paper

Troubleshooting Your VoIP Ecosystem with Prognosis

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This white paper from Integrated Research discusses how managing the complete VoIP ecosystem with Prognosis helps deliver improved ROI for VoIP and reduces management costs.

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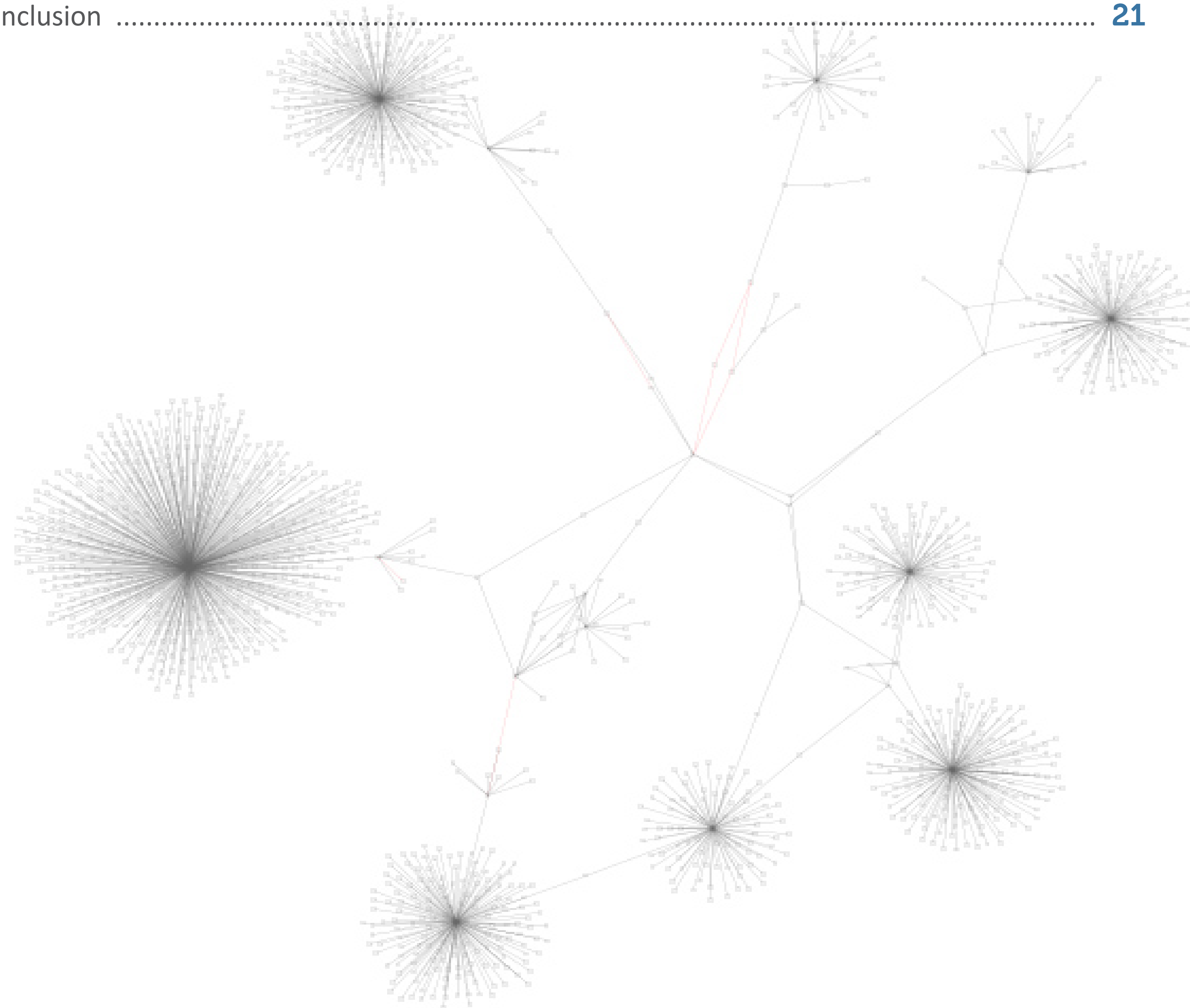

White Paper on a Page™

Top Issues

The VoIP call quality issues explored within this white paper include those within the network, within the VoIP application and within devices in the voice stream. Its aim is to help you as a VoIP stakeholder troubleshoot issues from a telephony perspective within the overall data network.

The top issues you'll need to address to ensure quality VoIP delivery include identifying slow network links and misconfigured or missing QoS settings. These often go undetected until VoIP is deployed, and then problems become apparent very quickly. Communications and lost productivity can result, and yet from the data network manager's perspective nothing is broken.

As VoIP delivery spans multiple operational domains, the top issues affecting call quality are often not caused by the application itself, but are related to its configuration, conflict with other applications for network resources and misalignment between VoIP and network design. The top issues are:

Poor Voice Quality

Slow network links delay voice traffic. A link can be slow for many reasons, and congestion can be one of them although not necessarily the only reason. Whatever the cause, a slow link can delay voice packet delivery resulting in poor voice quality. So what will help you identify slow network links?

Are your codec configurations suited to the available bandwidth and are your jitter buffer settings correct? Do you have enough prioritized bandwidth for voice traffic and sufficient bandwidth to cope with failed-over voice traffic on alternate routes? Prognosis monitors and measures voice streams and provides granular detail making it possible to analyze corresponding network performance on the voice stream path, in real time or historically. You'll then have the insights necessary to identify issues, develop work-arounds and resolve problems before they overly impact users and your business.

Dial Tone Failure

In the same way as network conditions can impact voice quality, failure to obtain a dial tone can be caused by network delays, poor hardware performance and outages. Phones may have become de-registered from the primary call server, backup devices may not be working properly or are not accessible, or download requests are failing when phones register. Your PBX may be severely overloaded and unable to cope with the demands of the busy hour.

How can you identify if these are at the root of the problem? Troubleshooting such a variety of causes is likely to incorporate handovers between operational domains which increases complexity, resources and time. Prognosis provides specialist VoIP monitoring, diagnostics and reporting for the entire VoIP ecosystem that helps you isolate and resolve problems to minimize the occurrence of VoIP issues bouncing back and forth between domains.

Call Setup Problems

Like dial tone failure, the complexity of troubleshooting call setup problems is compounded by the many interconnected components and domains the setup process spans. Successful call setup incorporates every component required to establish a path for the voice stream. Calls can fail to set up correctly if they cannot obtain a dial tone or access the PBX, voice gateways, the PSTN or locate sufficient trans-coding resources.

If your network experiences hardware failures without dial plan redundancy, gateways will not be able to set up calls to the PSTN. You'll also encounter issues if there are insufficient DSP resources for transcoding or conferencing, or location bandwidth limits are reached. Failure to configure alternate routing instructions and paths through the IP network, a downed gateway, device, network failure and excess bandwidth usage will also contribute to call setup failure.

Conclusion

Incorporating specialized VoIP ecosystem monitoring, alerting, management and reporting as part of an integrated management framework and defining efficient problem handover between operational teams will ease cross-domain problem management. Companies that pay attention to the complexities of VoIP management can significantly lower their annual operating costs and transform an old culture that reacts to problems into a new culture that solves problems before they escalate.

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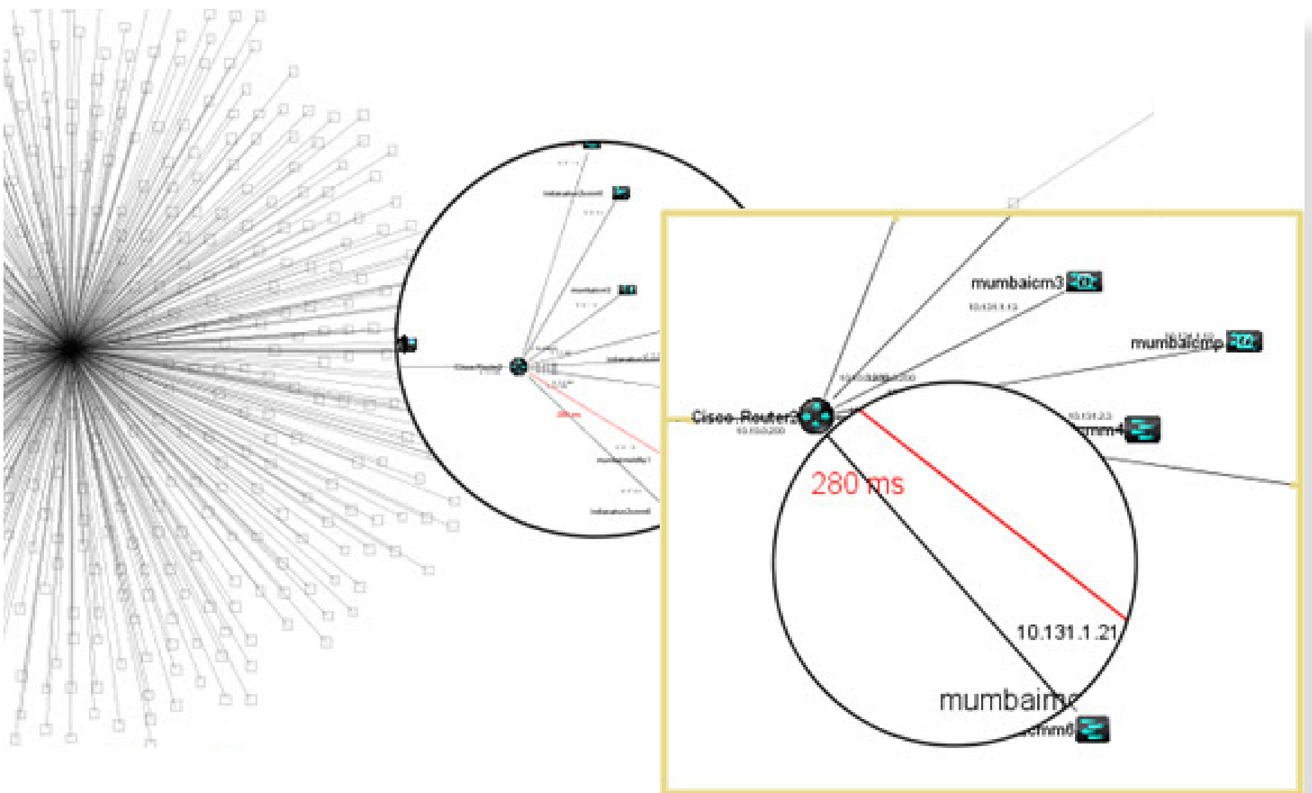
Part 1. Troubleshooting Poor Voice Quality

In the majority of cases, voice quality issues occur because of problems or conflicts within VoIP or between VoIP and other applications on the network. Part 1 of this white paper explores some of the most common causes of poor voice quality.

Slow Links Delaying Voice Traffic

Problem domain: WAN	Slow network links delay voice traffic. A link can be slow for many reasons, and congestion can be one of them although not necessarily the only reason. Whatever the cause, a slow link can delay voice packet delivery resulting in poor voice quality. So what will help you identify slow network links? A network map that is built from call activity and regularly updates link status will prove invaluable. As devices involved in the call are discovered, classified and monitored, slow links are identified, allowing you to focus on the devices involved in affected voice streams. You can then automatically correlate slow links on the network map with all devices involved in the calls. You can drill down to view details of the actual voice streams, and compare the latency, packet loss and jitter experienced on any call.
Most likely when: Early days of rollout and where QoS settings are not in place	Once the voice streams are identified, granular detail is needed to analyze corresponding network performance on the voice stream path, in real time or historically. This will then provide the insights you need to review QoS settings for any link that is experiencing latency and resolve voice quality issues.



VoIP ecosystem network view with slow links shown in red

Insufficient Prioritized Bandwidth for Voice Traffic

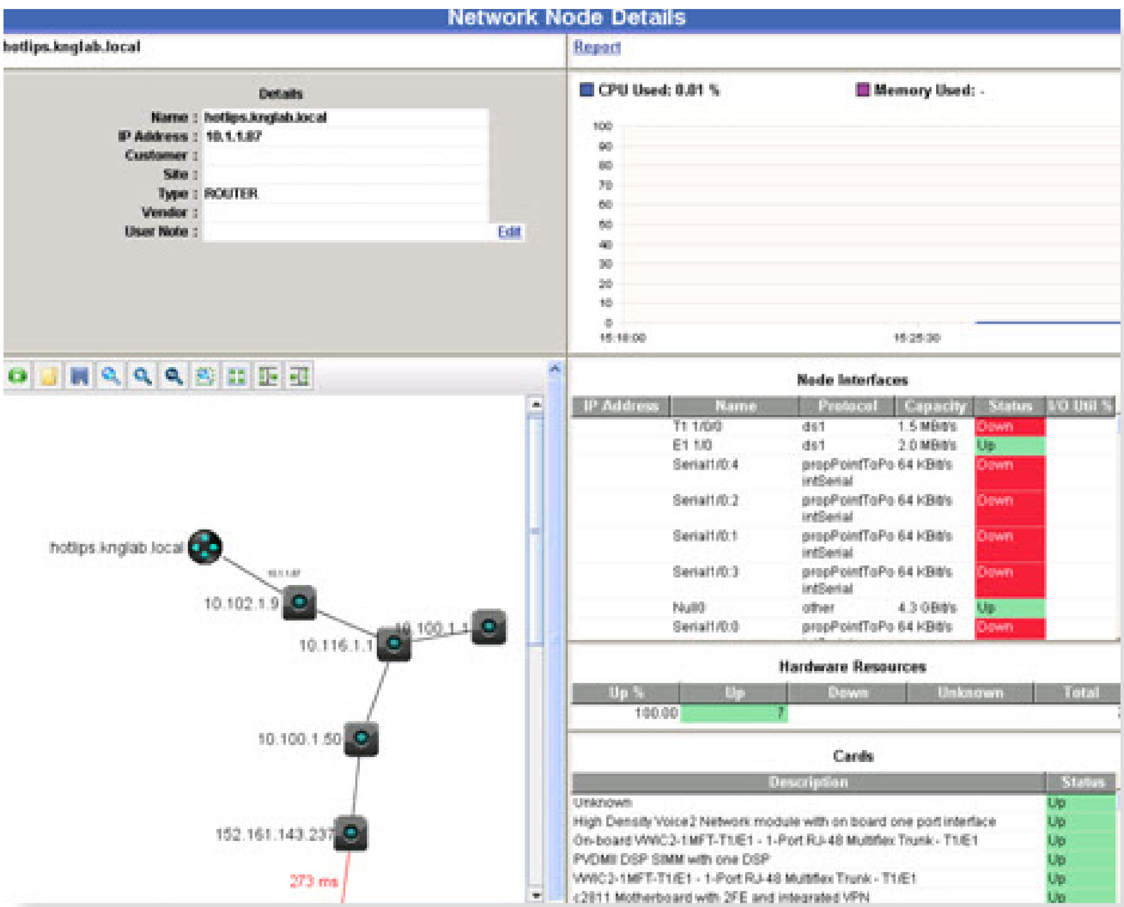
Problem domains: LAN/WAN	If insufficient WAN bandwidth is available, the addition of voice traffic is likely to result in general network congestion. This can cause link latency with voice packets arriving late, out of order, or being discarded, all of which will impair voice quality. As a lack of bandwidth is frequently perceived as the primary challenge to voice quality, the attempted remedy is often to purchase more bandwidth in small incremental chunks. However, this approach may still not accommodate the delay-sensitive, real-time demands of voice. A better approach is to measure existing call patterns to determine what capacity may be required before adding more bandwidth. Your carrier may be able to provide this information to assist you with establishing capacity to support busy hour requirements and the use of the WAN to route calls to PSTN gateways and other locations.
Most likely when: QoS incorrectly configured	You can also monitor VoIP network usage and obtain voice quality statistics including jitter, packet loss, latency and codecs and identify the calls experiencing quality impairments. You can use a network map to show the network mesh between VoIP devices and view full drilldown details for them, including node interface status, capacity and I/O percentage utilization. You can select information about the devices and their interfaces, the links between them and the VoIP streams they're carrying.

Switch/Router (Mpls/Diffserv) Configuration Errors

Problem domains: LAN/WAN	Data traffic prioritization techniques are only effective in helping achieve good voice quality if implemented on every switch and router in the voice path. A mis-configuration of even a single router in just one direction can cause delays that can cause impairments.
Most likely when: QoS incorrectly configured	Specialist VoIP ecosystem management provides immediate benefits for troubleshooting these types of configuration problems. Firstly from a telephony perspective you can view details of only degraded calls. Once you know exactly where the degraded calls are occurring you can focus on just the problem areas. You can then assess the potential MOS impairment factors, such as latency, packet loss, jitter codec and codec mis-configuration. Review displays that show you the operational status, ingress, egress and I/O percentage utilization of the devices and their interfaces so configuration errors can be determined and resolved.

Device Failure on Primary Route with Insufficient Bandwidth on Alternate Route

Problem domain: WAN	If a device fails on the primary route and insufficient bandwidth is available on the failover route, the addition of failed-over voice traffic is likely to cause route congestion. This will result in a noticeable impact on voice quality and means that other applications also using the alternate route may be affected.
Most likely when: Router failure has occurred	To assess the impact of a failover, active testing, or network assessment – can be used to check the performance of VoIP with other application traffic on both primary and alternate routes. By simulating packets generated from voice devices such as IP phones, voice gateways and voicemail servers, assessing engineers can observe how the network responds to variations in voice traffic patterns. They can then determine what network adjustments should be made to accommodate failover, usually in terms of QoS settings and link capacity. VoIP ecosystem management provides invaluable data when troubleshooting device failures. Firstly you'll receive a notification when degradation or failure occurs through customizable alert thresholds so you won't be subjected to alert floods. Secondly you'll be able to drill down and view network interface details providing information about the links to and from the interface such as IP addresses, utilization, duplex mode, link type and capacity, customer and site identifiers and any packet loss sustained. You can assess voice quality and VoIP performance over the alternate route and improve its capacity for future requirements.



Network node details

Excessive Bandwidth Utilization to a Particular Location

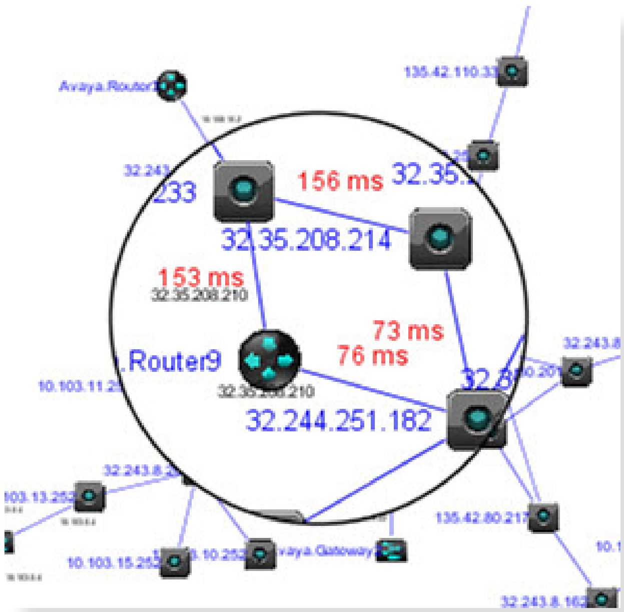
Problem domain: VoIP	Even if the overall VoIP service is performing well, a variety of factors can cause bandwidth over-subscription of a particular WAN link to a location. VoIP configuration errors, location bandwidth limits or excessive gateway use can all contribute to excessive bandwidth utilization.
Most likely when: Codec configurations unsuited to available bandwidth	If calls use a high bandwidth codec, all available capacity to a particular location may quickly be consumed. This will result in poor voice quality for all calls to and from that location and can also negatively impact the performance of other applications. To prevent this happening, codecs should be configured for each link to suit the available bandwidth. This might also require a resetting of user expectations of quality when placing long distance calls or setting service levels. Viewing detailed hop information helps you identify the busiest hop and the worst hop in terms of MOS cost. You can see the codec being used, the media encryption type and the value of the DiffServ traffic class. The accompanying network map provides drilldown details for routers and gateways for the associated voice streams. This information enables you to establish the throughput of the link, the codecs in use and the impairment factors being experienced.

Inadequate Jitter Management in the Network

Problem domains: VoIP, device	The fundamental way packets are sent across an IP network means that not all voice packets for a conversation are guaranteed to take the same route. Network congestion and jitter can cause voice packets to arrive unevenly or out of sequence, disrupting the conversation and causing garbled sentences.
Most likely when: Jitter buffer settings are incorrect	Jitter buffers on routers, gateways and end-user devices need to be configured with the correct delay setting. If it's too short – for example 10 ms or less – packets might be discarded needlessly. Since delays of up to 50 ms aren't detectable by the human ear, a higher delay setting of 30–80 ms is suitable for the jitter buffer to re-sequence packets. Prognosis displays the amount of jitter experienced on the call and the impact per call jitter has on MOS. It also breaks down the per-hop MOS cost of jitter relative to latency and packet loss. This allows administrators to determine the course of action for resolution. For example the MOS cost of jitter may be greater than latency so it makes sense to investigate jitter buffer settings before looking at overall ingress and egress utilization of the router.

Location Bandwidth Limits

Problem domains: WAN/device	Voice quality will suffer if there is over-subscription of available bandwidth to a particular location. VoIP users will experience poor voice quality as voice streams compete with each other and other types of traffic for bandwidth on the location's link.
Most likely when: Insufficient prioritized bandwidth to location	Bandwidth to a specific location can be protected by configuring a destination from within the VoIP application as a discrete location. Call admission controls can then be applied to protect voice traffic from the negative effects of other voice traffic and to reject calls when the volume exceeds specific thresholds.



Various locations experiencing link latency

Failure to Configure a Gateway as Part of a Location

Problem domains: VoIP, device	The WAN link to a gateway can become over-subscribed by callers outside its location. This can happen when a dial plan routes every call for a dialed sequence to a single gateway. As a result, voice quality on all calls will suffer as too many calls are routed to the gateway's location.
Most likely when: Gateway is not added as a location's resource	A real-time network map built from active calls will make this problem easy to identify as it will display all the calls attempting to use the gateway. Administrators can then see the impact this has on the gateway's interfaces. A way to manage this problem is to configure the gateway as part of a VoIP location. This allows you to include it in call admission controls for the location. In this way, voice quality on a WAN link to a particular location is protected from the negative effects of additional voice traffic. Analyzing the voice stream details shows you all the devices in the voice stream path and all links experiencing latency. Correlating this with router interface and hardware resource information lets you see the available capacity compared to utilization, as well as the status, duplex setting, and protocol in use.

Summary

In the majority of cases, voice quality issues occur because of problems or conflicts within VoIP or between VoIP and other applications on the network. Insufficient prioritized bandwidth, slow links, router configuration errors, device failures and location bandwidth limits can all affect voice quality.

Specialized VoIP monitoring, VoIP ecosystem diagnostics and reporting help identify the root causes of poor voice quality. Fine grained telephony-specific metrics, real-time network maps, deep diagnostics and voice quality reporting can identify the stream or streams experiencing degraded quality.

Once the voice streams are identified, granular detail makes it possible to analyze corresponding network performance on the voice stream path, in real time or historically. This will then provide the insights necessary to identify, develop work-arounds and resolve issues before they overly impact users or the business.

Such insights are vital to successfully managing the performance of voice as a business service over an IP network.

Part 2. Troubleshooting Dial Tone Failure

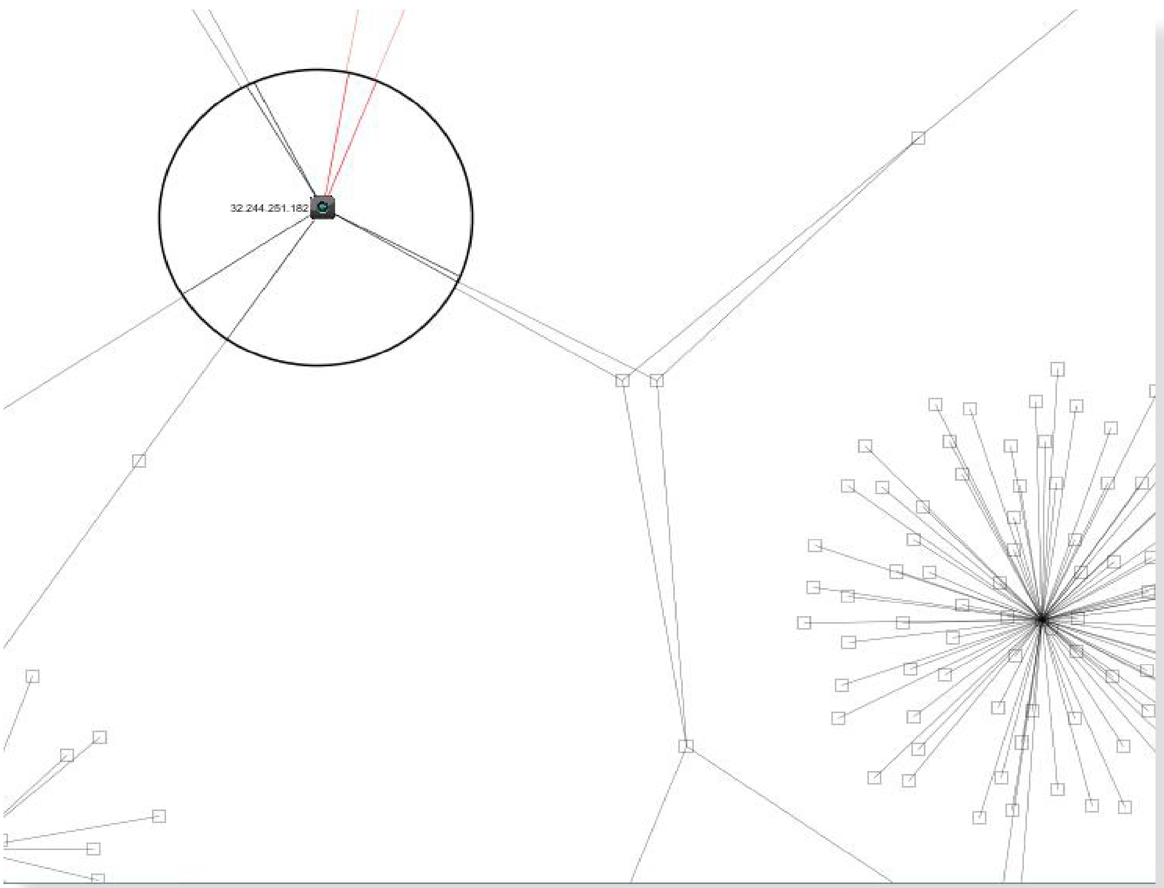
In the same way as network conditions can impact voice quality, failure to obtain a dial tone can be caused by network delays, poor hardware performance and outages. Part 2 of this white paper explores some of the most common causes of dial tone failure.

Phones De-Registered from Primary Call Server

Problem domains: Device, LAN/WAN, VoIP	Phones de-registering from a primary call server or IP-PBX can be caused by software, hardware or digital signal processor problems on the phone, network failures, slow links, configuration errors or software bugs.
Most likely when: Hardware, software or network failures occur	To help identify if the source of the problem lies with the phone, call server or network, it's useful to establish if the phone or the call server supplies dial tone. By viewing the relationships between the phones and the PBX and categorizing them by type you'll know if they rely on the PBX for dial tone or use another mechanism, like a SIP phone, to provide it themselves. This allows you to establish a starting point for troubleshooting. If the phone is functioning correctly or a replacement phone exhibits the same problem, then troubleshooting needs to extend to the data network. Integrating a VoIP ecosystem map with VoIP management provides an at-a-glance view of the entire VoIP ecosystem. This enables you to see if links are experiencing latency that can contribute to phone deregistration or failure to obtain a dial tone. Using this information, you can confirm that critical servers are available to the phones and that their IP addresses are valid. If the phone is able to reach these servers on the LAN or WAN, you can check that the phone has the correct firmware version. Failure to register with any call server typically indicates a broader network problem and this can introduce the added complexity of a domain handover. The problem is manifested in the VoIP application, but it is core network configurations and infrastructure that must be investigated. You may need to work with the data network manager to confirm DHCP servers are functioning properly with correct security settings and are able to issue IP addresses that are not blocked by firewall or router settings. Also, you may need to investigate faulty network modules or foreign exchange station cards that may cause silence or absence of dial tone in a voice port. Specialized VoIP management is very useful in troubleshooting dial tone failure because it clarifies the relationships between individual components and calculates their status based on their inter-dependencies. For example a trunk includes a gateway, boards, ports and route patterns. If a board is down but the other components are up, the trunk is considered degraded rather than down. However, if all the components are down then the trunk itself is considered down.

Failure of Backup Device

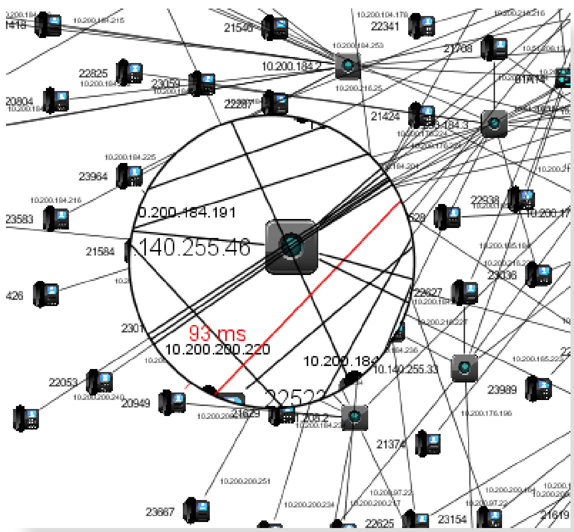
Problem domains: VoIP, device	Failure to obtain a dial tone from a backup device may be caused by the device itself, as well as by the inability of phones to reach and register with it. One cause of this failure could be a slow link between the phone and the backup device.
Most likely when: Hardware or network failure, or configuration errors	An integrated VoIP network map will show you where the problem link or links lie and show VoIP data contextually with server performance metrics. This will enable you to correlate network performance with VoIP quality. If phones are attempting to fail over, it's essential to ensure that backup devices are available and that they are accessible across the LAN or WAN. Troubleshooting should include checking phone and backup device status, failover configurations and network link status and speed.



Slow link to backup device

Download Requests Failing when Phone Registers

Problem domains: Device, LAN/WAN, VoIP	Even phones that supply their own dialtone require access to various download servers via TFTP or HTTP/S, (depending on vendor implementation) for firmware, software updates, configuration information and locale data.
Most likely when: Problems with availability, accessibility and configuration of download and call servers	If requests to provide scripts, applications, images, settings, boot files or firmware are failing, it is advisable to check server availability, capacity, performance and security. You'll need to ensure that the phones are able to reach the required servers, that the correct file versions are being downloaded and that phones requesting the files have sufficient permissions to receive them. Latency can cause download requests to fail or time out. Establishing link latency, analyzing devices and their interfaces along the download path will help identify and resolve the cause of failed download requests.



Slow link between phone and download server

Severe PBX Overload

Problem domains: Device, LAN/WAN, VoIP	A severely overloaded PBX or call server will be unable to service all the requests it receives, including supplying dial tone. The PBX call load should be monitored to identify the busiest hour and the most common daily busy hour. Checking capacity and performance of the call server at these times will identify when it becomes overloaded and potentially unable to supply dial tone.
Most likely when: Insufficient IP-PBX capacity	If busy-hour data is sourced from multiple PBXs, it'll be necessary to calculate busy hour statistics for each PBX individually, as well as aggregate it for all PBXs. This will determine how far the telephony system is pushed at its peak. Specialized VoIP monitoring, diagnostics and reporting provides details on call throughput, call attempts, call failure and success rates and CPU utilization. This information provides invaluable data for troubleshooting, capacity planning, PBX consolidation and failover contingency planning.

Summary

Dial tone failure can be caused by phones deregistering from the primary PBX or call server, network or server latency and failures, and overloading of the PBX. Troubleshooting such a variety of causes is likely to incorporate handovers between operational domains which increases complexity, resources and time.

To help reduce complexity, specialist VoIP monitoring, diagnostics and reporting for the entire VoIP ecosystem provides voice stream information from both telephony and network perspectives. This helps isolate and resolve problems to minimize the occurrence of VoIP issues bouncing back and forth between domains.

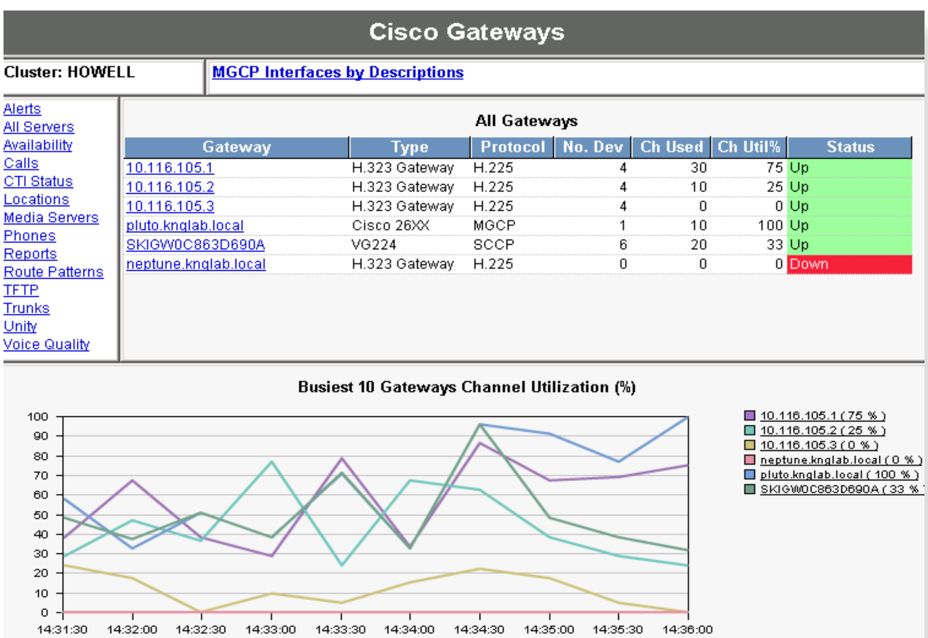
Another benefit of VoIP ecosystem management is unified information presentation. This enables the easiest correlation between application and network performance. You can replay network and application data collections and compare them to gain a full picture of trends in the network over time.

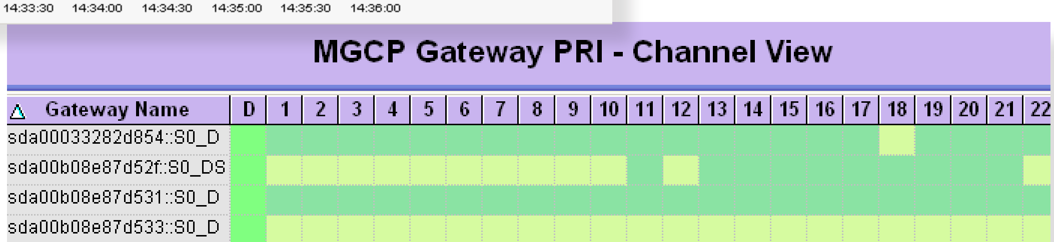
Part 3. Troubleshooting Call Setup Problems

The complexity of troubleshooting call setup failures is compounded by the many interconnected components and domains the setup process spans. Calls can fail to set up correctly if they cannot obtain a dial tone or access the PBX, voice gateways, the PSTN or locate sufficient trans-coding resources. Calls will also fail to set up if there are long delays that cause users themselves to abandon the attempt. This final part of the white paper examines the most common root causes for call setup failure.

Gateway Interface Going Down

Problem domain: Gateways	A gateway cannot set up calls to the PSTN if there is a hardware failure or insufficient allocation of redundant channels and trunks to a dial plan. This will become a serious issue if redundancy is not configured in dial plans or if all redundant routes are down or busy. If the gateway appears to be down it is advisable to check configuration, capacity and consumption of PSTN resources.
Most likely when: Hardware failures and no dial plan redundancy	Even if the local gateway is functional, users may receive a fast-busy signal due to network congestion somewhere outside your control. If the call setup relies on local exchange carriers, any congestion within their network may cause call setup attempts to fail. Specialized VoIP ecosystem management provides feedback on gateways' operational status and generates alerts if they degrade or fail. This identifies impending problems, allowing investigation and resolution before users are affected. Using a real-time VoIP network map enables you to visualize potential trouble spots, and identify slow links to gateways. You can then focus on just that part of the network and drill down to the gateway's status, capacity, activity and utilization.





Insufficient Dsp Resource for Transcoding or Conferencing

Problem domains: Device, LAN/WAN, VoIP	Increased call setup times and failures can result from insufficient digital signal processor (DSP) or mislocated trans-coding resources. As resources that handle conferencing and codec conversion are usually located in pools, careful thought needs to be given to their placement.
Most likely when: Insufficient or mislocated DSP resources	If calls need to be bridged and/or conferenced it's important that DSP resources are in the correct location. This prevents calls traversing the network, consuming bandwidth, increasing setup time and decreasing resource availability for other users. VoIP ecosystem management enables you can analyze channel status and activity showing the DSP's status and the number each DSP is processing together with voice stream details of users on conference calls. This information can be used to determine if additional channels need to be brought on line, consolidated or optimized.

Bandwidth Limit Reached to Destination

Problem domains: LAN/WAN, VoIP	If call admission controls are in place, new call setup attempts will be rejected when the bandwidth limit is reached to a destination location. This is done intentionally to prevent new voice streams negatively impacting active calls. Bandwidth use to a destination can be controlled based on the number of calls, or by bandwidth thresholds to ensure sufficient capacity is available.
Most likely when: Call admission controls in place or absence of call admission controls and excessive traffic on WAN link	Unless dial plan redundancy is configured, a user attempting to set up a call that exceeds call admission control thresholds will fail. The call will only be set up successfully once sufficient bandwidth is available to the destination. VoIP monitoring and analysis provide details of the number of calls that are being rejected due to bandwidth limits to particular locations. It also provides a detailed breakdown of calls by origin, routing, PBX or call server. This helps with capacity planning decisions to properly support call setup requirements. This may include increasing the percentage of existing bandwidth for calls, increasing overall bandwidth to the location or routing calls elsewhere via dial plan redundancy configuration.

Route Pattern/Dial Plan Congestion/Failure

Problem domain: Gateways	Dial plan congestion may be caused by insufficient gateway resources, a downed PRI, or call admission controls rejecting new call setup attempts. It can also occur if insufficient channels or trunks are allocated to the dial plan to accommodate the volume of calls.
Most likely when: Hardware failures and no dial plan redundancy	For example, a dial plan may try to use a remote SIP gateway. The PBX routes the call to the gateway but the gateway is down and does not respond. The specified alternate gateway is down and a third gateway has resources available but is in a location where call admission controls prevent the request from being serviced. All these factors result in the user's call setup being unsuccessful because the route pattern or dial plan cannot access the resources it needs. However, troubleshooting dial plan related call setup failures is particularly complex because the relationships between dial plans, route partition tables, route patterns and trunk groups are not always obvious. Specialized VoIP monitoring, network diagnosis and analysis is crucial. It provides route and dial plan information with deep drill-down diagnostics that help determine which trunks or gateways are causing the problems – and can dramatically reduce mean time to repair of service-impacting issues.

Users Abandon Calls Due to Long Call Setup Delay

Problem domains:
Device, LAN/WAN, VoIP

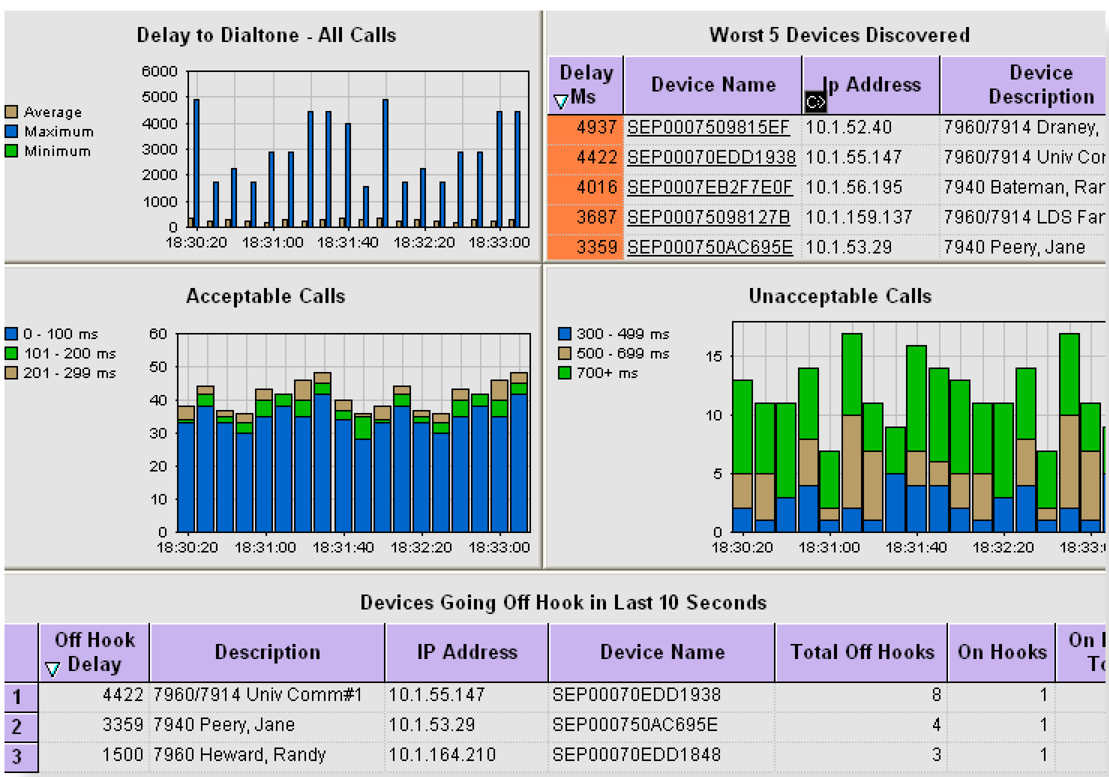
In traditional telephony, users are accustomed to hearing a ring-back tone well within five seconds of entering a number. If they have to wait longer they may lose patience and abandon the call. Default waiting periods for call signaling timers within VoIP introduce the potential for much longer delays. In fact, it's possible to wait up to 32 seconds for a ring-back tone, but from the application's perspective, nothing is wrong.

Most likely when:
Insufficient or mislocated DSP resources

As well as long default waiting periods in the signaling protocols, failure to hear a ring-back tone may also be due to unavailability of a DSP resource to play it, or voice quality issues between the DSP resource and the phone. This makes it particularly difficult to identify which calls are actually failing, compared to those that have been abandoned by the caller because they didn't hear a ring-back tone.

An integrated network map identifies slow links within the VoIP ecosystem. This will allow you to address latency as a potential source of long call setup delays.

You'll be able to determine the root cause by monitoring and reporting on call load and attempted, failed and rejected calls. This helps you identify which calls were successfully set up, which failed, and why.



Summary

Successful call setup incorporates every component required to establish a path for the voice stream. Dial tone failure, inability to access the PBX or route calls to gateways, the PSTN or failure to locate trans-coding resources are all reasons call setup attempts can fail.

In addition to these root causes, failure to configure alternate routing instructions and paths through the IP network, a gateway PRI going down, device, network failures and excess bandwidth usage will also contribute to call setup failure. Specialized VoIP management assists support teams with troubleshooting call setup failures by monitoring and reporting on every component required to successfully set up calls.

Conclusion

The top issues affecting VoIP call quality are often not caused by the VoIP application itself, but are related to its configuration, conflicts with other applications for network resources and misalignment between VoIP and network design. To effectively troubleshoot VoIP problems, you must monitor, measure and manage its availability and performance across all the operational domains it spans.

Incorporating specialized VoIP ecosystem monitoring, alerting, management and reporting as part of an integrated management framework and defining efficient problem handover between operational teams will ease cross-domain problem management. Companies that pay attention to the complexities of VoIP management can significantly lower their annual operating costs and transform an old culture that reacts to problems into a new culture that solves problems before they escalate.

And more importantly; VoIP ecosystem management reduces the instances of problems occurring over time and helps you achieve a more rapid return on investment.

VoIP ecosystem management procedures and features will depend on the platform being monitored. Contact us for more information.

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Integrated Research
Since 1988, Integrated Research has been providing Prognosis performance monitoring software solutions for business-critical computing environments. With offices in the USA, Europe and Australia and a global channel-driven distribution network, the company services customers in more than 50 countries.

Prognosis for IP telephony was launched in 2000, and has been chosen by many of the world's largest companies to manage their IP telephony deployments. These include the world's two largest aerospace companies, two of the largest US electric utility companies, three of the largest US financial services firms, and the largest telecommunications companies in the US, Germany and France.

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