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VoIP Advantages for the Contact Center

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Rev—170110

Synopsis

VoIP is a technology that can bring savings to the modern contact center in a variety of ways. It has always competed well with TDM on a minute by minute price basis. But more interesting, VoIP allows the contact-center the ability to do new applications services that simply are not possible to implement in traditional TDM circuit space. This article explores several of those "VoIP advantages" and attempts to show what is possible using VoIP SIP Trunking in a modern contact center.

It's now the 21st century and IP networks have long since surpassed TDM networks in the amount of raw data that is being transported at any given moment throughout the world. The reasons for this are many, but in the short we find packet technology simply adapts better at transporting varying types of media than TDM technology does. Packet networks are also far easier to maintain. As a result natural selection has packet networks winning the evolutional battle over TDM networks and technologies.

Synopsis (Continued)

With VoIP, the new feature functions for voice, and specifically voice to and from the contact-center, are creating new possibilities on a level not seen since the digital TDM T1 circuit was introduced.

This paper will explore several of those VoIP advantages for the contact center. Specifically, we'll focus on:

- Freeing Inbound Traffic from geography
- Dynamic Traffic Control
- Call Re-direction and Routing

In so doing this paper should enlighten the reader to the possibilities of what could be done in a modern contact center using VoIP technology at its core.

The TDM Contact Center

Today's TMD based contact center is setup up with circuit connections to one or more long distance providers, and to an incumbent LEC if inbound local DID services are required. These circuits are dedicated to that given TDM carrier. As a result, in many cases this means the contact center has to deploy T1 circuits to many different carriers to get the overall needed blend of services; i.e. separate T1 circuits for outbound LD, inbound 800, inbound DID, etc.

Further, these T1 circuits are hard-wired to the TDM carrier's switch at the other end. That results in a very static network topology that offers lots of limitations, and some very serious single points of failure.

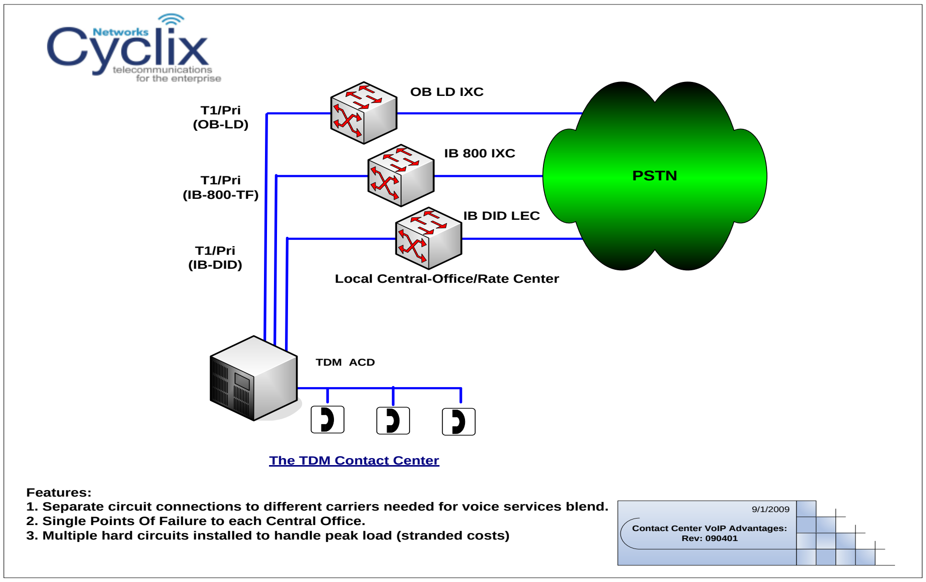

Fig. 1: The TDM Contact Center

Freeing Inbound TF and DID Service From Geography

Exploring these TDM limitations further, let’s examine inbound DID traffic on TDM networks to the TDM contact center, and the numbers assigned to those services. In the TDM world any given contact center is limited to using DID numbers in the region it is located. The local central office, and its associated PSTN rate center, determines what that number is.

Such a situation forces a very limiting scenario on the contact-center by their TDM carrier for inbound DID services. Unless expensive point-to-point leased circuits are employed, there is no flexibility here to make the contact center appear to be in any other geographic rate center market other then what it physically is. That is a major problem, and an inherent design flaw built into today’s incumbent PSTN architecture.

But with VoIP, and SIP Trunking, we fortunately have a much different situation. This "geographical limitation" is completely eliminated. DID calls to a contact-center connected via VoIP are still transferred through the traditional TDM PSTN rate center. But once the call passes through that rate center, and moves into the IP realm, it is freed from its circuit restrictions.

Freeing Inbound TF and DID Service From Geography (Continued)

Once in IP space the call can be terminated anywhere the end user wants it to go. There are no limitations. The call can be transported to any signaling endpoint location on that IP network, anywhere in the world, with virtually no routing restrictions or price penalties. As such geographical limitations are removed and the call in the IP domain achieves what Cyclix terms “Global Number Portability (GNP).”

In the case of Cyclix Networks being the ITSP, once the call is in the IP domain it is routed to Cyclix Network’s ENUM routing Server™. Cyclix then routes and forwards the call across the Cyclix network, terminating it to wherever the client has defined the SIP endpoint to be. This location is determined by the customer, and not the phone company!

With this concept of “Global Number Portability”™ end point routing change orders now also become trivial. If the customer wants to change to a different geographical location, it requires little or no configuration on their part, and is generally instantaneous.

So back the contact center… by deploying VoIP technology in the contact-center it is now free to establish as many DIDs in as many geographic rate centers as it chooses, yet deploy physically in only one or more locations. It’s entirely up to the contact center designer, not the phone company, where to locate, and how many locations to deploy.

Summarizing, **the ability to free telephony from its PSTN rate center, and the inherent geographical ties, is a major technological step for the contact center.** This feature alone will be one of the key factors that spur future mass migration away from TDM networks to packet networks for voice telephony in the contact center.

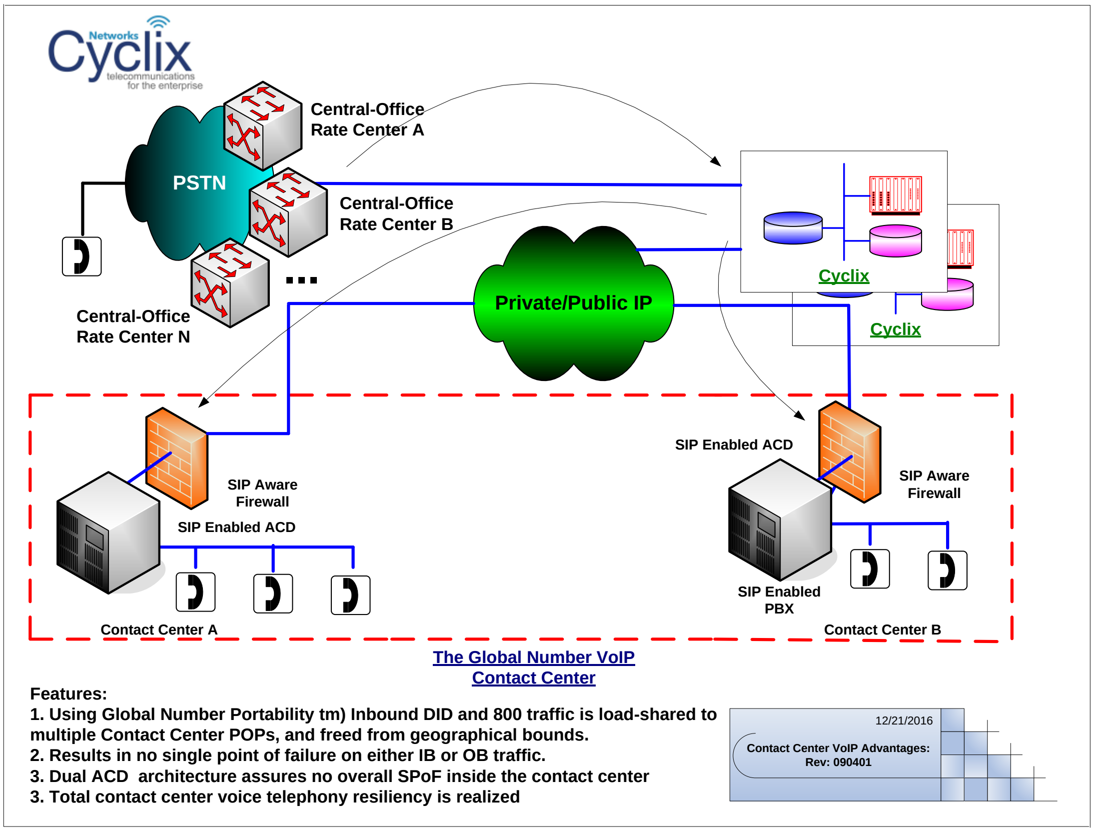

Fig.

2: The Global Number VoIP Contact Center

Handling Peak Traffic Loads

In TDM space one must have a set amount of circuits connecting the contact center to its carrier's TDM switches. The amount of circuits one must provision is a function of what max call load the user wants to be able to handle at peak load periods.

This presents a major problem for the contact center designer, as most of the time capacity will only be at 30%-50% level, yet then spike for some brief moment during the day, such as when an add campaign runs on mass media, like radio or television.

Handling Peak Traffic Loads (Continued)

This dilemma for the contact center manager having to handle this peak load is it requires a build-out of circuits to handle that maximum traffic point. Then off peak those extra circuits sit idle. But unfortunately the monthly re-occurring invoice for those mostly idle circuits does not sit idle. There is a stranded charge for those relatively inactive and unused TDM circuits levied on the contact center client.

The fundamental problem here is that circuit-based TDM technologies consume resources whether they are being used or not. Those circuits, tied to TDM central office switches, consume ports on those switches, and those ports must be paid for, regardless of whether they are used for traffic or not.

VoIP, on the other hand, takes a radically different approach. There are no ports consumed because there are no physical circuits, only virtual circuits. Those virtual circuits can then be added or taken away dynamically as needed.

Further, the IP marketplace bills for bandwidth used, not circuits. That allows one to scale up to meet peak traffic periods by provisioning a bandwidth burst option; i.e**. Companies pay extra when during peak traffic bursts, but do not pay for the extra bandwidth when their traffic goes back to baseline.**

This ability to have a burst payment option for voice bandwidth gives VoIP a significant pricing advantage over TDM circuits, as when the bandwidth is not needed, it is also not invoiced. Further, this radically simplifies the overall network design, as the transport mechanism become truly dynamically scalable, easier to manage, and so, saving further on support and manpower costs.

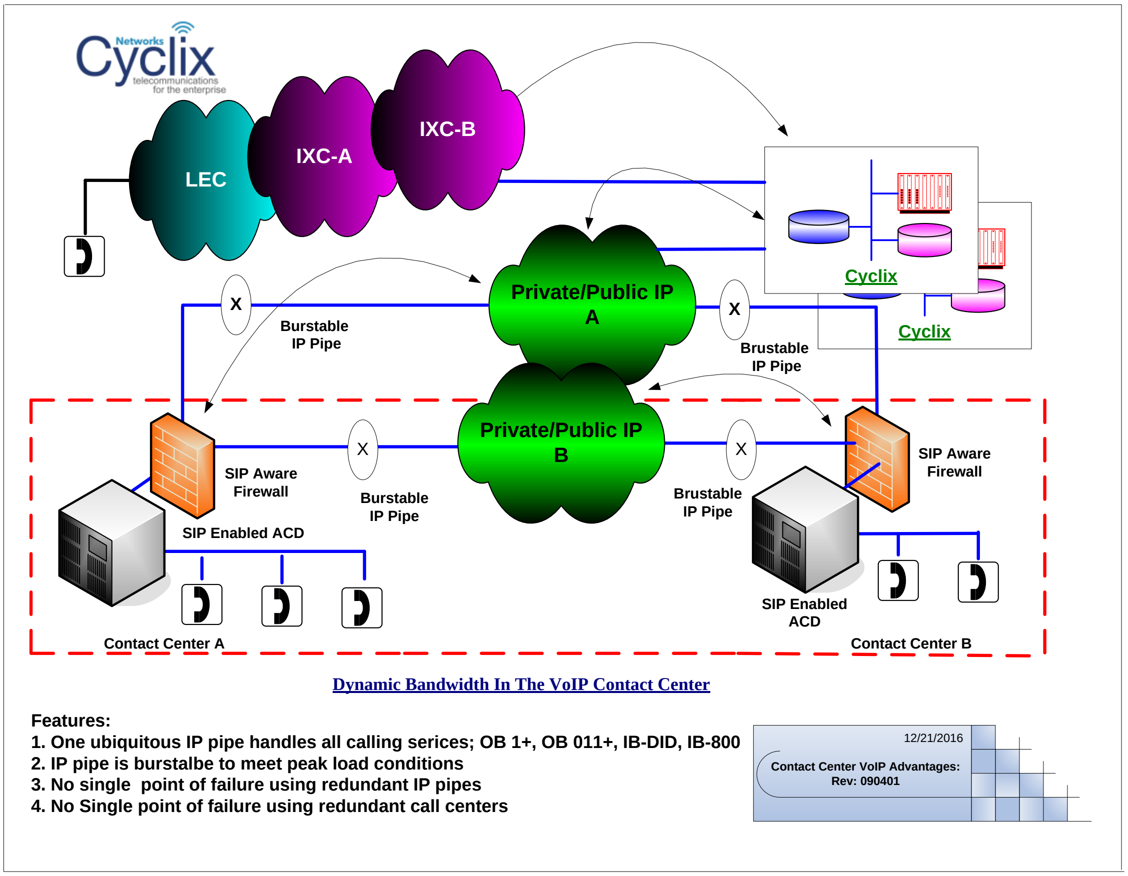

Fig. 3: Dynamic Bandwidth in the VoIP Contact Center

The Freedom to Re-Direct

In this day and age national contact centers tend to be 24x7 enterprises which cannot afford to go down, as else the business they support is down. Therefore, as a general rule, a contact center wants a level of redundancy that will guarantee survival, and not have a single point of failure. Operations at 99.999% uptime 24x7 hours a day, 365 days a year is the goal.

How does one achieve such a high-reliability goal in TDM space? Not easily. TMD circuits once again are forcing the end user to hardwire voice connectivity to a specific carrier's switch. That switch then becomes the life support system for the contact-center. It's an inflexible arrangement that provides a single point

The Freedom to Re-Direct (Continued)

of failure, and furthermore is rigid and closed in how it allows the end user to control re-routing of its traffic in the event of emergencies or shutdowns for maintenance.

Specifically, with 800 toll free traffic on the PSTN, routing to end points is done via a device called the SCP (Service Control Point). This is a platform deployed in the PSTN carrier's SS7 signaling backbone. It acts as the central control point for all the carrier's inbound 800 traffic to the contact-center end user.

SCP's are closely guarded devices due to the enormous responsibility they have in controlling the TDM carrier's overall 800 traffic flow. As a result very few carriers have opened up these platforms to the end user for applications such as emergency re-routing, or even basic load sharing between contact-center sites.

At best LD carriers might offer the ability for the customer to phone in an 800 re-route. But these orders then execute in carrier time, not real-time. So there is very little flexibility offered here in PSTN space for 800 Toll Free traffic coming into the contact center.

For inbound DID traffic, the situation is much worse. PSTN DIDs are hard coded to hit a specific PSTN rate center. As a result, there are no options to do redirection of the call in the TDM network. Therefore the only thing an end user can do is field their own switch, then hairpin calls to another location if a re-direct is required. But now we have an extra call leg charge, and a TDM switch to maintain at the customer's premises. The cost and complexity of doing this is generally unworkable.

Fortunately (again) the IP scenario is much different. In an IP environment the contact-center is not connected to a single voice switch fabric device per se. Rather, the Network itself is the switch, as all the needed addressing info accompanies each voice packet that flows into and out of the contact center.

As a result this "address with data" architecture offers us some major advantages over TDM technologies. The contact-center user can simply re-direct calls in real-time for whatever reason, whenever, and without picking up any cost penalty if the calls stay on-net in IP space.

The Freedom to Re-Direct (Continued)

As an example, if an enterprise ran two separate call centers, and one needed to go down for maintenance, the incoming SIP/VoIP calls could simply be re-directed to the other VoIP based ACD at the other call-center. The calling party would not know any redirect was taking place, and the contact centers would not incur any additional charge, as the call stayed on-net in IP space.

Load sharing and intelligent routing applications are also much easier to do in VoIP environment on the ITSP carrier side. Cyclix has taken the SCP concept outlined earlier and built control point functionality that is made available and accessible to the end user. This allows call center managers to control traffic into their own sites in real-time, be it for load sharing between sites, hot standby failure protection, or whatever.

Further, all this new traffic routing control in VoIP space applies to DID traffic, not just Toll Free 800. With this change, DID traffic has become totally dynamic and re-routable on demand in the VoIP domain. This is something that is simply impossible to do in traditional TDM circuit space.

Summarizing, inbound call redirect routing functionality is either too costly, or in the case of DID, impossible to achieve in TDM space. But a contact center deployed with VoIP now has routing options open up to it that are unheard of in TDM space, and essentially unlimited. This gives the VoIP contact center a significant competitive advantage over the TDM contact center.

Conclusions

VoIP technology, and specifically SIP Trunking with Cyclix, is revolutionizing the possibilities for the contact center and help desk. Freeing voice from “the circuit” allows new possibilities that are simply impossible or too costly to implement in legacy switched TDM networks.

Summarizing some of those benefits we’ve covered here are…

- **Freeing Inbound traffic from Geography—**With VoIP Inbound toll free and DID traffic we are now free to terminate to any SIP IP contact center endpoint, turning what was a totally fixed static model into one that is now user definable (not carrier defined), user configurable, and totally dynamic.

Conclusions (Continued)

- **Handling Peak Traffic Loads—**In IP space you pay for only what you use. You do not pay for what you do not use. That makes VoIP a far more economical way of dealing with the problems of traffic spikes and peak load situations. TDM technology simply cannot compete with VoIP in this scenario.

- **The Freedom to Re-Direct—Once a voice call is moved off the PSTN and into IP space with VoIP it is free of the circuit and the PSTN rate center. That freedom allows new options and applications for the contact center pertaining to redundancy, load sharing, and traffic re-direction that are simply impossible to duplicate in TDM networks due to cost and or inherent technical limitations.**

Certainly, VoIP offers some up front price advantage over traditional TDM telephony on a minute by minute comparison. But the technical advantages that VoIP brings to the table are of a quantum leap in nature. VoIP will ultimately allow the contact center to achieve new levels of service, reliability, and feature/function that are simply not possible with today's (or shall we say yesterdays) traditional circuit technology solutions.

So simply stated, VoIP is a sea change technology for the contact center. It is only a matter of time before it becomes the core mechanism of transport for the industry. Those that understand the advantage, embrace it, and adopt it will flourish. Those that do not will be left behind, stuck in a dead end circuit model that will limit their ability to compete.

To explore Cyclix VoIP services further for your contact center, please contact Cyclix sales today at:

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- www.cyclixnet.com

Also see the white paper section on our website (www.cyclixnet.com) for further discussions on technical issues pertaining to VoIP, and how it relates to the next generation VoIP enabled contact center.