



# Codec Normalization for Wholesale Voice Termination Providers

## Ditech Platform

Ditech's Codec Normalization solution is available on the Packet Voice Processor.



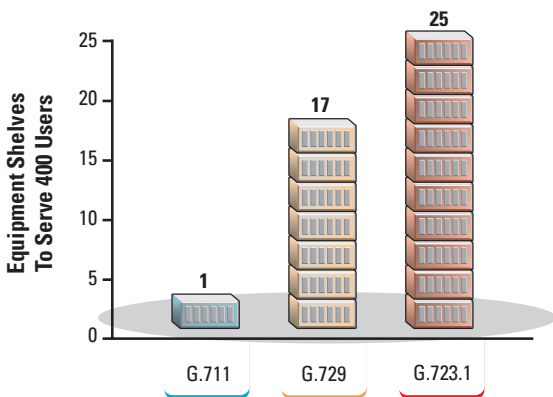
**Packet Voice Processor™**  
1000Base-T/SX/LX

Telephone calls using Voice over IP (VoIP) are becoming more commonplace as service providers and content creators develop new VoIP applications. However, VoIP is far from universal – it still requires interconnection to the Public Switched Telephone Network (PSTN) to complete most calls. Because many early adopters of VoIP do not have their own PSTN, carriers offering wholesale VoIP termination and origination services have become increasingly important. For these service providers, a major challenge has been the choice coder/decoder (codec) to offer on the VoIP side. The PSTN uses G.711, but for many VoIP applications, G.711 is not the best option. Advancements in codec technology have created the possibility of using different voice codecs for specific technical and business-related applications. For example, bandwidth constraints within access networks may require low bitrate codecs, whereas networks with more bandwidth may choose to differentiate themselves by offering wideband codecs for better-than-PSTN voice quality. Access networks requiring PSTN termination can be very diverse, requiring the use of many codecs including G.711A/μ, G.726, G.729a/b/e/g, G.722, G.723.1, EVRC, GSM-HR, GSM-FR, iLBC, and Speex.

## The Need for VoIP Codec Normalization

As PSTN termination and origination services grow, operators must decide how they begin to support a diverse customer base (many codecs) with their VoIP service, while still providing an economical offering. Today, support for additional codecs requires support for each codec on all nodes in the call path. From an operations point of view, this can be a significant commitment to test, deploy, and administer. From a capital standpoint, the costs

become prohibitive as the network grows and each supported codec adds cost to the media gateways, media servers, and application servers. Not only is the addition of a new codec expensive, but it also reduces channel density because of the increase in MIPS required by more complex codecs. For these reasons, providers often have to deploy additional equipment to accommodate new codecs (Figure 1).



A typical conferencing application where each shelf is capable of 400 MIPS.

Figure 1 :: Equipment Required by Various Codecs

## SUPPORTED CODECS

- G.711 μ-law
- G.711 A-law
- G.723.1
- G.726
- G.729
- G.729a
- G.729b
- G.729ab
- G.729e
- G.729g
- iLBC
- G.722\*
- G.722.1\*
- G.727\*
- G.728\*
- GSM-HR, FR, EFR\*
- GSM-AMR\*
- G.722.2 (AMR-WB)\*
- VMR-WB\*
- EVRC, EVRC-B, EVRC-WB (4GV)\*
- Speex\*
- BroadVoice® 16/32\*
- iSAC\*
- iPCM-wb\*

\*Future Capability

## The Codec Normalization Solution

The most straightforward way to increase the number of supported codecs is to perform codec normalization at the border of the VoIP network. This allows a diverse set of application-specific codecs in the access network to operate with a single codec in the core. The codec in the core remains G.711 because it requires the lowest processing effort, has a reasonable cost-to-density ratio, and is universally supported by the various network elements.

Ditech Networks' Packet Voice Processor™ is an ideal solution for supporting diverse codecs in the access network. The Packet Voice Processor provides codec transcoding for up to 13,440 channels in a shelf, and over 40,000 channels per telco rack. It is available as a fully redundant, NEBS-compliant chassis for central office deployment.

The Packet Voice Processor offers significant savings compared to media gateway or media server based solutions, where additional codec options can increase equipment cost by as much as 75%. Furthermore, the cost-effective architecture of the Packet Voice Processor allows selective allocation of DSP resources to only those calls requiring treatment, helping to reduce overall network costs while still maintaining high network quality. The benefits of the Packet Voice Processor include a reduced codec burden on application and feature servers and the ability to add or upgrade codecs without affecting equipment in the core.

For codec normalization, the Packet Voice Processor is located at the network border as shown in Figure 2. At the boundary between the aggregation/access network and the core VoIP network, the Packet Voice

Processor normalizes all media traffic to a specific codec since all media must traverse this aggregation point. The core VoIP network can then consist of equipment that only supports one or two key codec types (e.g., narrowband G.711 and wideband G.722), while the access network can support any codec type that is desired.

## Conclusion

Implementing Ditech Networks' codec normalization solution at the VoIP border allows for a wider set of customers to connect to the wholesale network without major changes to the network. Having a consistent interface inside the wholesale network reduces equipment costs, improves interoperability, and allows network growth to occur quickly and cost-effectively.

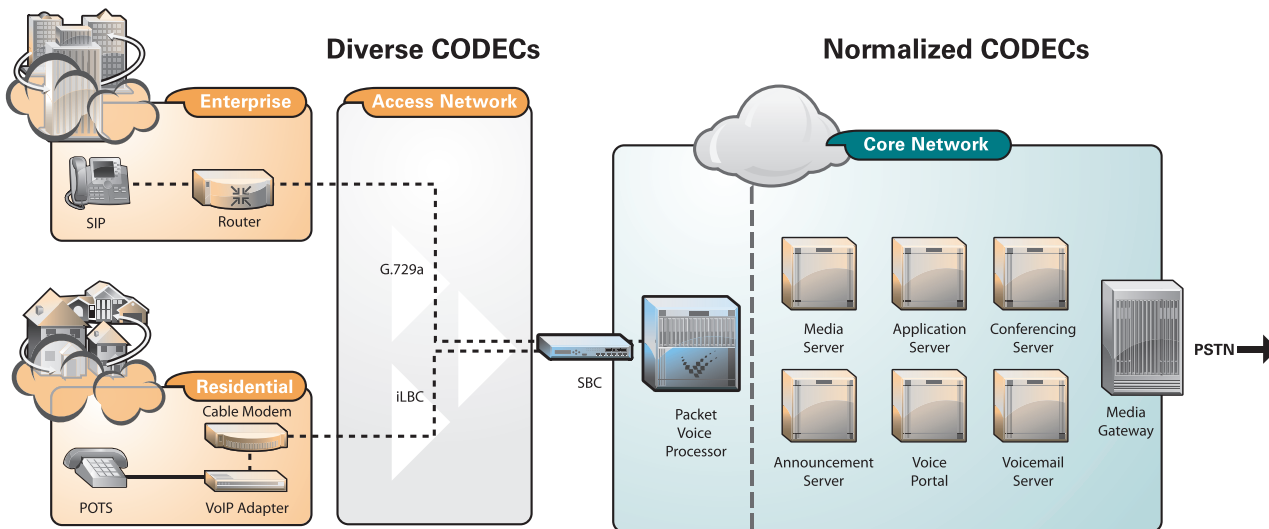


Figure 2 :: Ditech's Codec Normalization Solution at the VoIP Border



**Ditech Networks**  
825 East Middlefield Road  
Mountain View, CA 94043  
USA

800 234 0884 toll free  
800 770 0117 support  
650 623 1300 direct  
650 564 9599 fax

[ditech@ditechnetworks.com](mailto:ditech@ditechnetworks.com)  
[www.ditechnetworks.com](http://www.ditechnetworks.com)