Ditech Networks

Expanding Codec Support for VoIP Conferencing Services

Ditech Platform

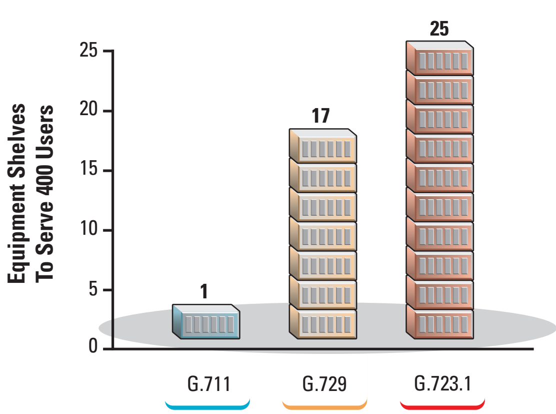
Ditech's Codec Normalization solution is available on the Packet Voice Processor.

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

The ability for Voice over IP (VoIP) to deliver advanced collaboration services is driving its adoption with conferencing providers. However, as VoIP conferencing applications become more prevalent, customer experience can be degraded due to the appearance of many new problems. A major concern is the coder-decoder (codec) used to convert analog voice to digital. Traditional telephone networks use G.711, but with many new VoIP services, G.711 is not optimal. Advancements in codec technology open the possibility of using a variety of voice codecs to solve specific technical and business problems. Since most conferencing services do not offer dedicated network access, customers may try to connect using a variety of codecs including G.711A/μ, G.726, G.729a/b/e/g, G.722, G.723.1, EVRC, GSM-HR, GSM-FR, iLBC, or Speex, to name a few.

The Need for VoIP Codec Normalization

As conferencing moves increasingly toward the VoIP domain, conferencing operators are facing a dilemma. How do they support a diverse customer base (i.e., many codecs) on their application while still providing an economical, high-quality service? Today, support for additional codecs requires support for each codec type on all nodes in the call path. From the operations point of view, this can be a significant commitment to test, deploy, and administer. From a capital standpoint, the costs can become prohibitive as the network grows because specific codec additions add cost in the media gateways, media servers, and application servers. Not only is the addition of a new codec an expensive proposition, but it also reduces channel density because of the increase in MIPS required by more complex codecs. For these reasons, most providers have to deploy additional nodes of the same equipment (Figure 1), increasing their total capital expenditure.



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 conferencing network. This allows a diverse set of application-specific codecs in the access network while using a single codec in the conferencing network. The codec for the conference network and bridge remains G.711 because it requires the lowest processing effort, has a reasonable cost-to-density ratio, and has universal support from other 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 may be somewhat limited and 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 development burden on application and feature servers, reduced operational expense, and the ability to add or upgrade codecs without affecting operation of existing equipment in the core.

For Codec Normalization, the Packet Voice Processor is placed at the network border, as shown in Figure 2. At the boundary between the aggregation/access network and the conferencing network, the Packet Voice Processor normalizes all media traffic to a single codec since all media must traverse this aggregation point. The conferencing 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

Ditech Networks' Codec Normalization solution provides operators with significant benefits by establishing a consistent codec interface between users and the conferencing network. This consistent interface reduces equipment costs, improves interoperability, and allows network growth to occur quickly and cost effectively.

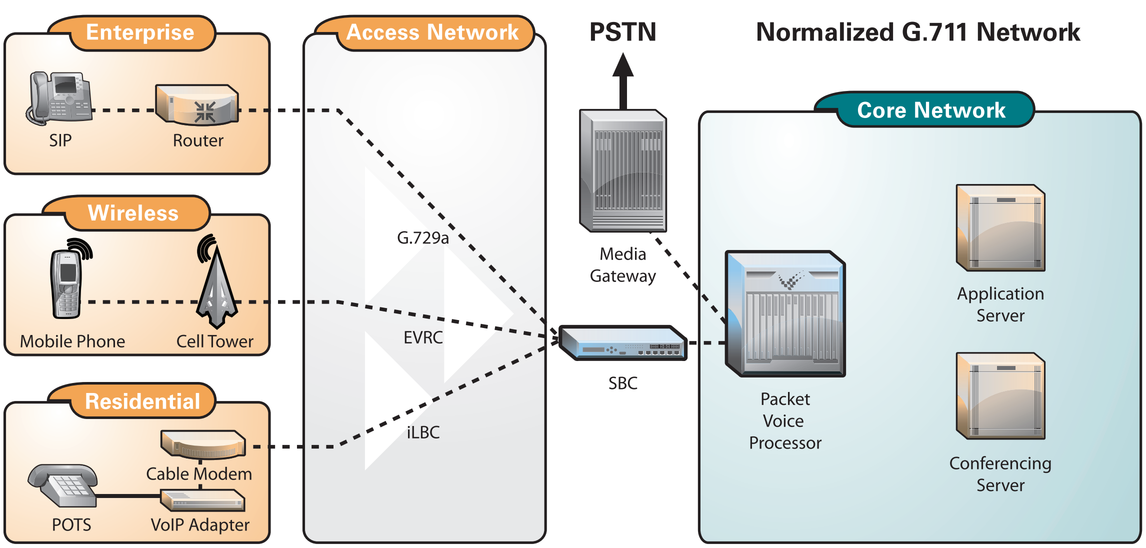


Figure 2 :: Ditech's Codec Normalization Solution at the VoIP Border

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
www.ditechnetworks.com