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Codec Engineering: Guidelines for Designing Narrow-Band, Wide-Band and Low-Bit-Rate Codecs into International Voice Networks

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international ip interconnection

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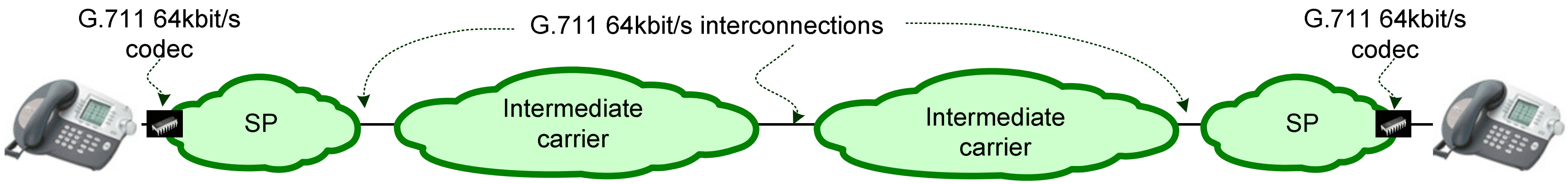
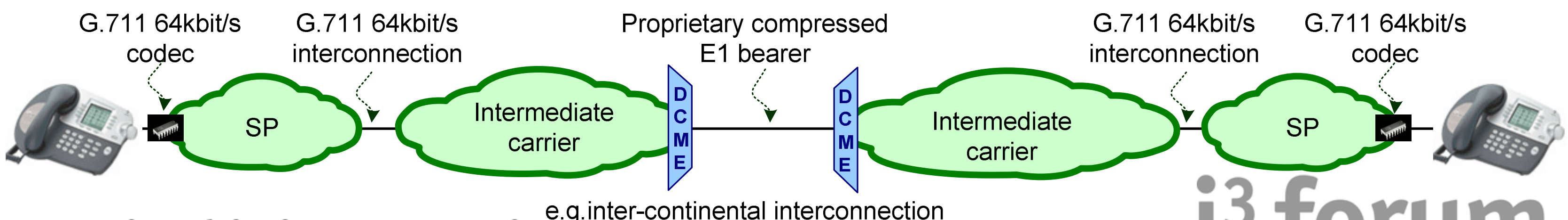
Why Are Codecs a Call Quality Issue with International VoIP Networks?

- TDM networks designed for 64kbit/s speech
  - G.711 codec universally used in fixed networks
  - Mobile networks - G.711 is interconnection standard

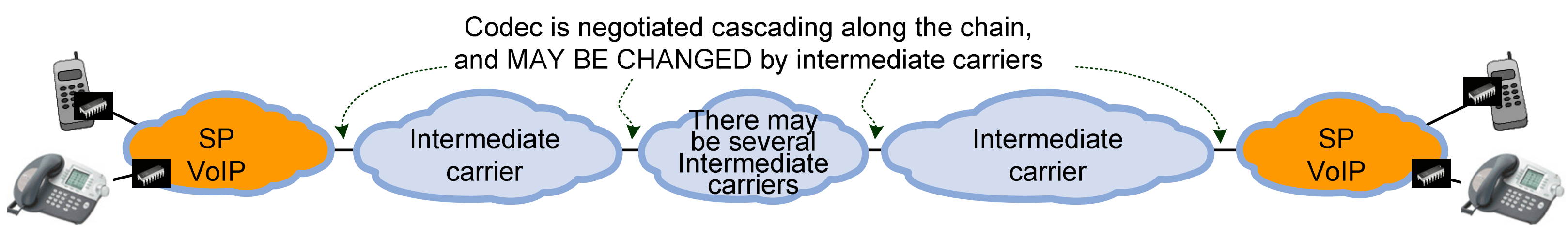


- For inter-continental international networks, "voice compression" often used:
  - DCME transcoded to a lower bit rate codec and suppressed transmission of silence
  - Average voice transmission rates achieved
    - 16kbit/s (G.726 codec)
    - 8kbit/s (G.728 codec).
  - Always the switch interface was G.711 64kbit/s



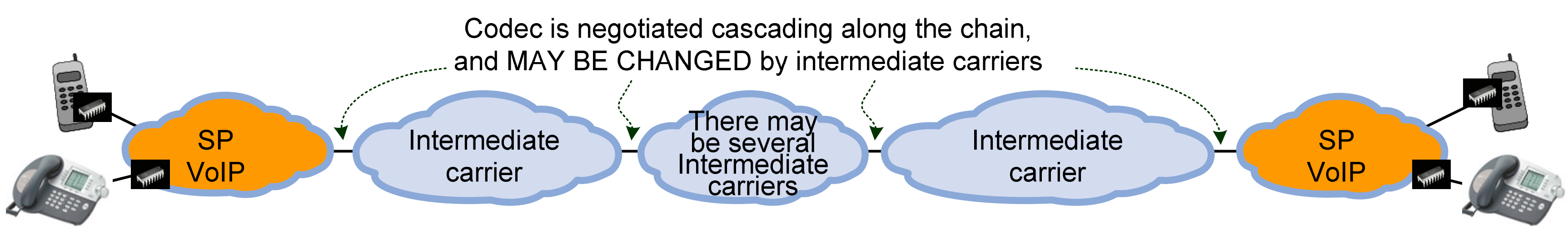
Why Are Codecs a Call Quality Issue with International VoIP Networks?

- VoIP has no common codec standard,
  - more codecs now available,
  - more being invented
- Codecs remain the primary responsibility of Service Providers
- Due to diversity of networks and codec choices in the World,
  - transcoding <u>will occur on international calls</u>
  - voice quality <u>will be impacted</u>
- codec is part of the call set-up negotiation



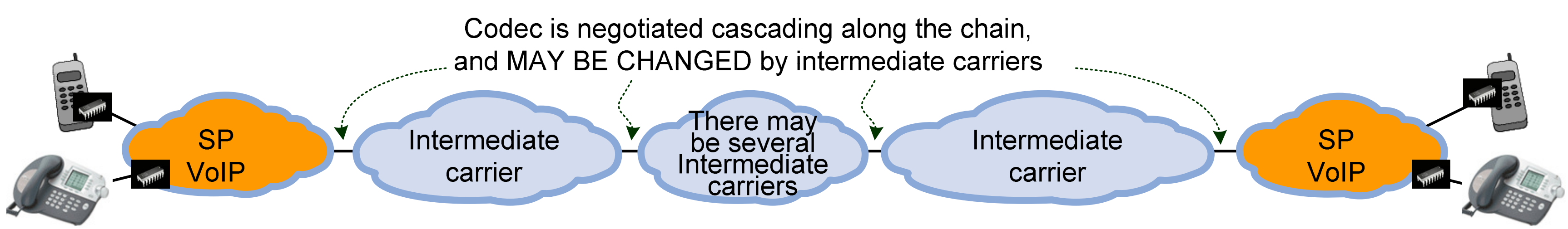
The Problem of Many Codecs

- Voice Engineering
  - Intermediate (usually international) carriers
    - may not have visibility of the end SP's
    - visibility is of the ingress and egress carriers networks only
    - may not know if transcoding has already occurred,
  - International carriers are the greatest affected by codecs (and IP transmission latency) but have little control over these design variables



End - to - End View is Required

- End – to – End, NOT Local
  - lowering voice bit rate within a carrier network to save bandwidth looks attractive
  - This is WRONG EMPHASIS, leads to
    - lost $$$ because of customers disliking lower call quality
    - damages carriers reputation
  - End-to-end (meaning ear–to–mouth) approach must be taken
    - This includes cordless phone codecs!



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Economic Impact of Poor Call Quality

- A good (natural) voice call means customers linger and just talk.
- Low voice quality means customers hang up early (revenue lost)
  - may "sink" deals won on basis of high quality parameters
  - loss of customers to competitors
  - extra load on call centres

Main Impact

MOSCQE (See later slide)	R-Factor (see later slide)	% of customers terminating call early	Loss on 3B minutes pa @ 1c/min margin, assuming ALOC is halved for customers terminating early
3.74	73	1%	$150,000
3.6	70	1.7%	$255,000
3.1	60	6%	$900,000
2.58	50	18%	$2,700,000

Simple impact of customers hanging up early

Codec Types and Call Quality

- Narrow band codecs (speech 300Hz – 3.4KHz)
  - PSTN standard bandwidth
  - low tolerance to transcoding and latency
- Wideband codecs (speech 50Hz – 7KHz)
  - improve speech quality
  - higher tolerance to transcoding
  - offsets the problems with NB
  - BUT new, and will take time to evolve into ALL networks
- LBR codecs (Low Bit Rate Narrow Band codecs)
  - required when low transmission rates are necessary because bandwidth is scarce or expensive,
  - lowest tolerance to transcoding and latency

Standard Framework for Call Quality Planning – Voice Quality Measurement

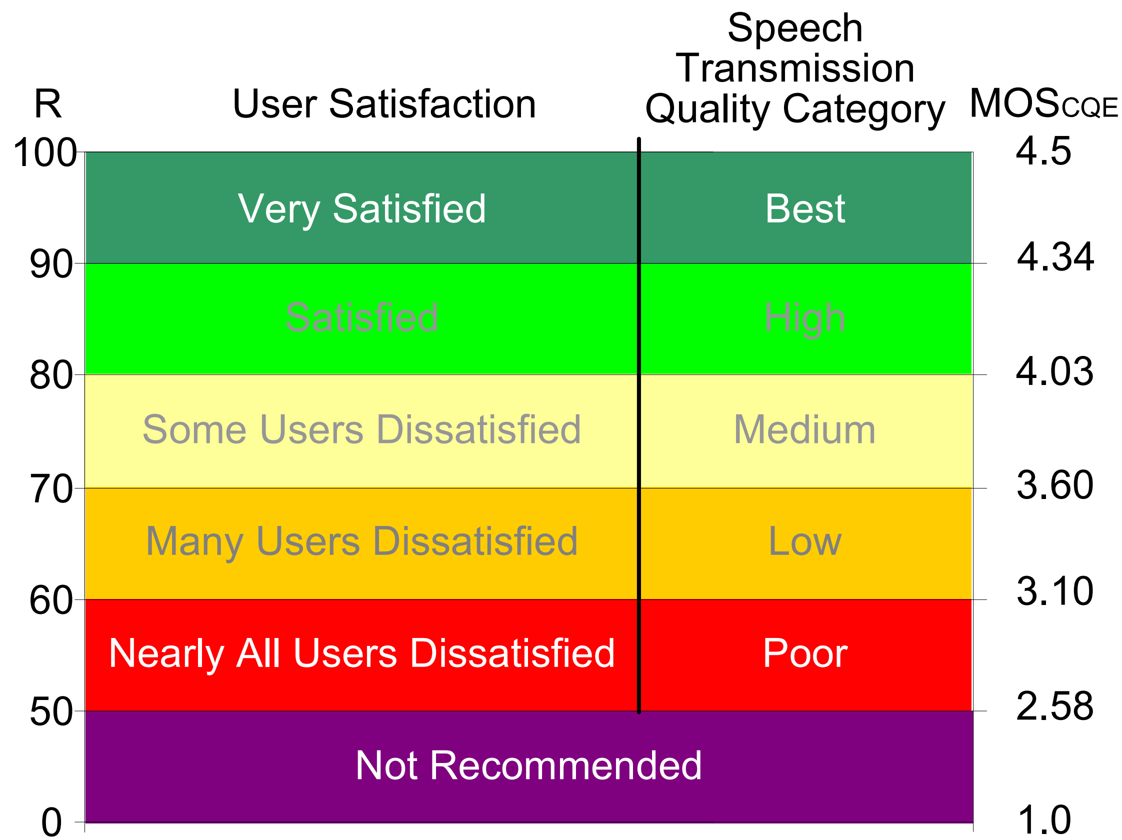
- Measuring Voice Call Quality
  - ask people to classify calls to a MOS scale (Mean Opinion Score).
- MOS is
  - a customer choice,
  - its what customers actually think,
  - is the absolute reference.

MOS	Classification
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

Scale of MOS values.

Standard Framework for Call Quality Planning – A Narrow Band Voice Quality Planning Tool

- Voice Quality predicted using an ITU-T developed E-Model
- E-model output is
  - R-Factor
  - used to predict MOS scores (called MOS_{CQE})
- R-Factor calibrated in Voice Quality Bands (although Quality Scale <u>is a continuum</u>)



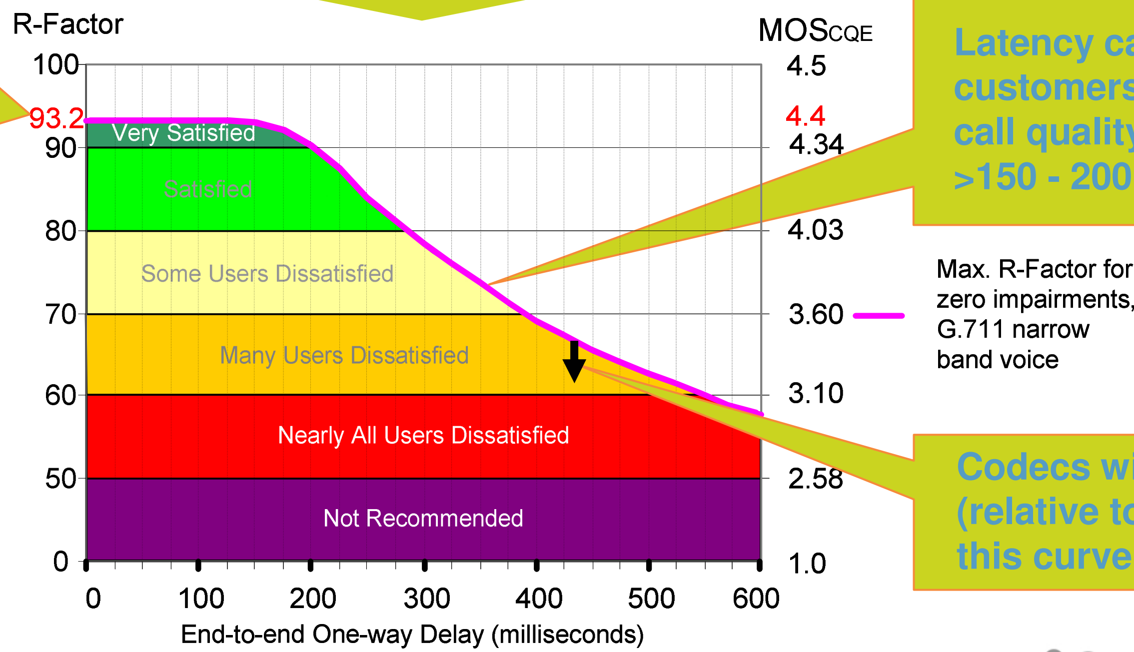
Standard Framework for Call Quality Planning – A Narrow Band Voice Quality Planning Tool

- E-Model: ***R-factor* = *Ro* - *Is* - *Id* - *Ie* + *A***

Impairments due to: Noise | Loudness, sidetone, echo etc | Delay | Codecs | Advantage Factor

- Factors sum but some have to be solved for graphically
  - *(e.g. Delay and echo)*

G.711 64kbit/s is defined at R=93.2 (MOS testing)

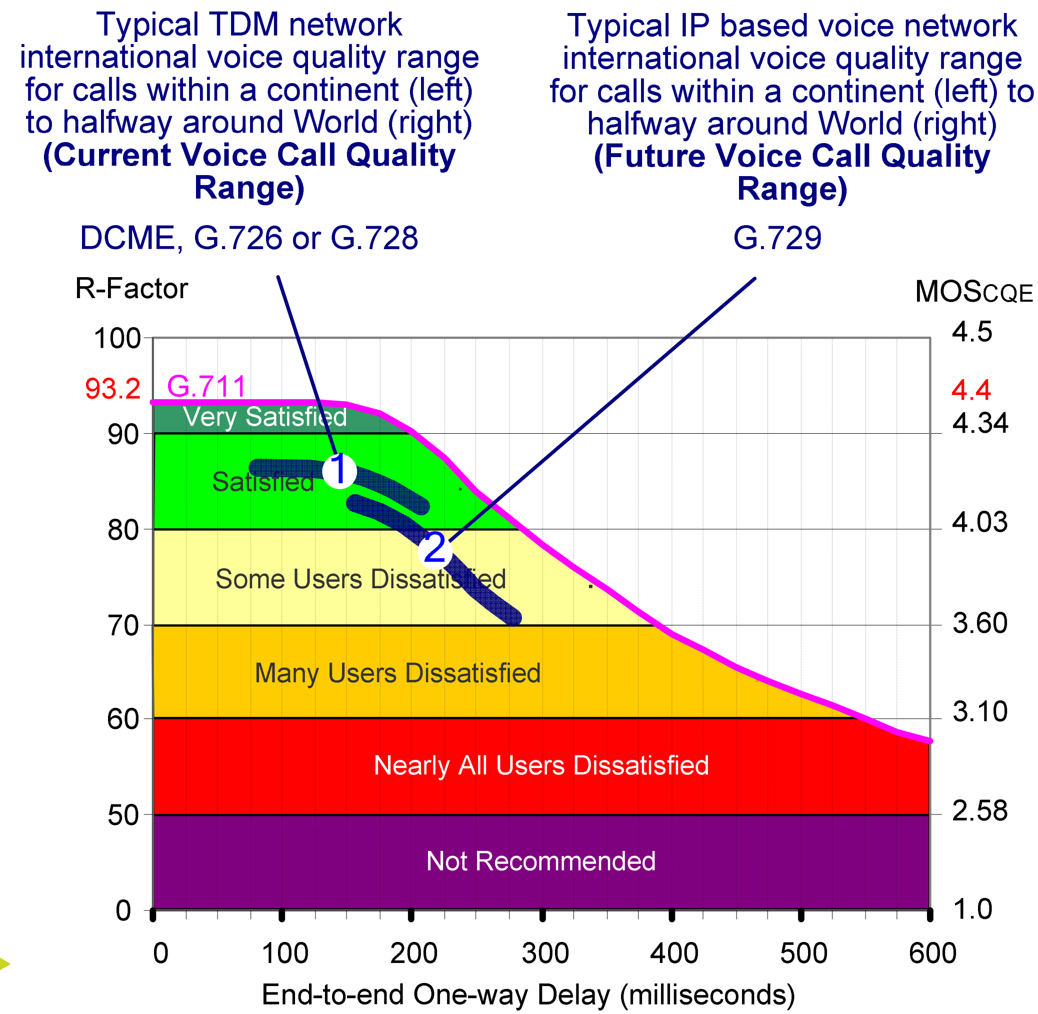


Latency causes customers to perceive a call quality decrease >150 - 200ms

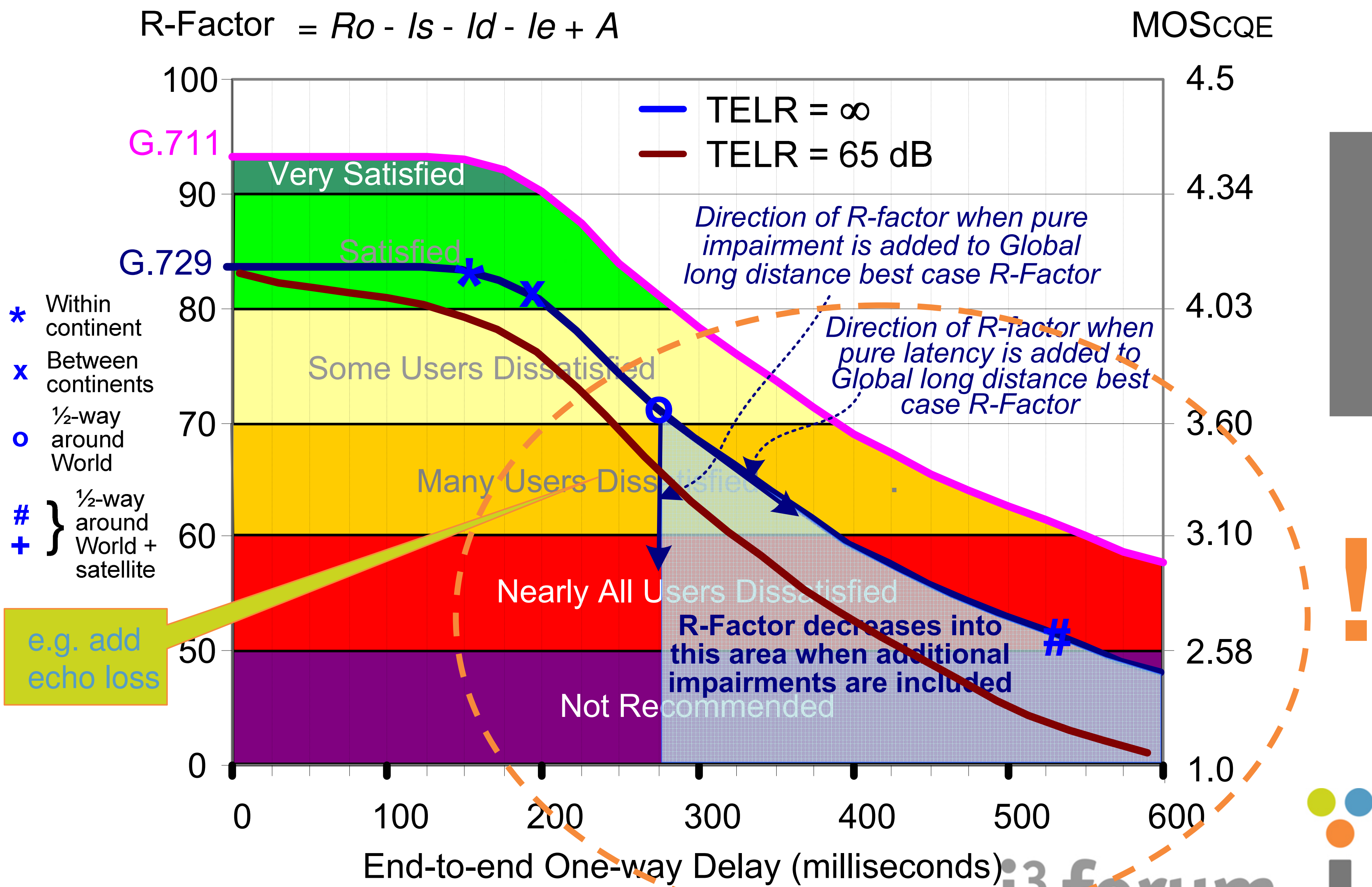
Codecs with Higher *Ie* (relative to 93.2) lower this curve

Illustrating the VoIP Voice Quality Problem –
Narrow Band Codecs (300Hz – 3,400Hz)

- TDM
  - Lower latency
  - generally low impairment, low latency codecs
  - delivers generally satisfied international voice customers
  - DCME example ► 1
- Converting to VoIP
  - packetisation latencies
  - generally higher impairment, higher latency codecs,
  - drives lower customer satisfaction ratings for equivalent calls
  - e.g. G.729 ► 2



Lower Tolerance for Additional Impairments

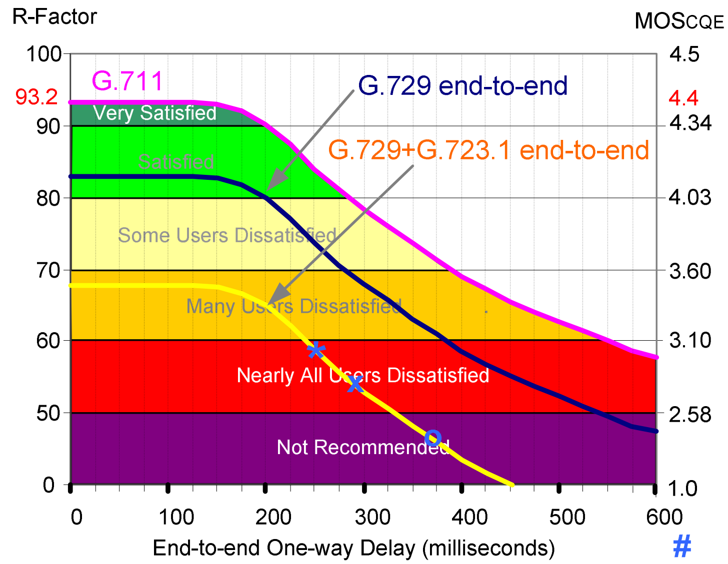


It Gets Worse Yet – Transcoding due to differing Codec Choices

Narrow Band Codecs (300Hz – 3,400Hz)

- 3 examples
- NB: codec impairments ONLY are shown
- See White Paper for full explanations (http://www.i3forum.org/sites/default/files/i3F%20-%20Technical%20White%20Paper%20on%20Codec%20Release%202%20(2010-May).pdf)

* Within continent
x Between continents
o ½-way around World
+ } ½-way around World + satellite



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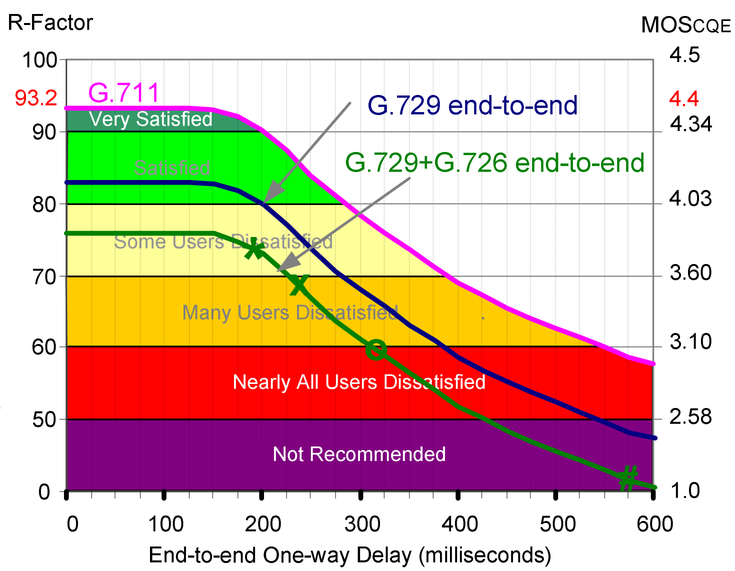
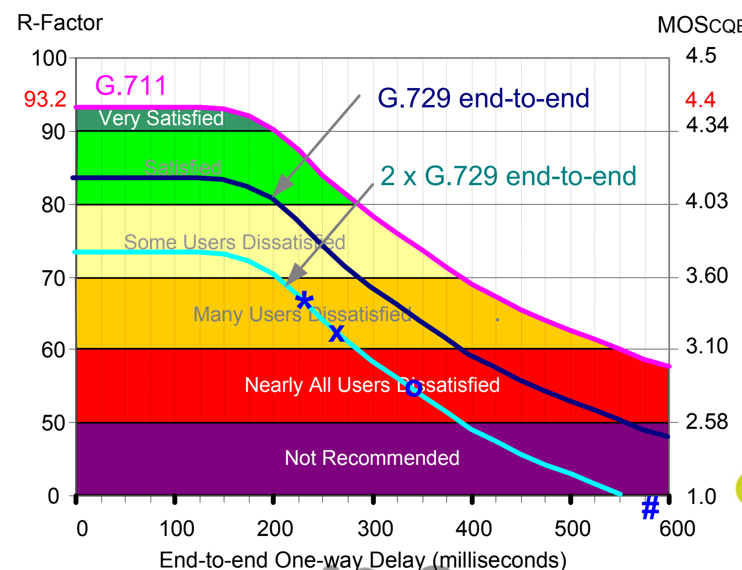




Wide Band Codecs - Help is on the Horizon

Deep Voices Crisp Sound

- Wideband Codecs (50Hz – 7,000Hz)
  - more natural sound
  - greatly improved sensation of presence
  - better intelligibility
  - superior listening comfort.
- Improved speech quality offsets the transcoding problems with NB transmission
- BUT as yet NO E-model available to predict wideband voice call quality overall, apart from
  - wideband codec impairments characterised *Ie,wb*
  - R-factor scale extended to R = 129
  - CANNOT mix NB *Ie* values in
  - no noise, echo effects etc can yet be modelled

Wide Band Codecs – When Will They Help ?

- Wideband codecs are
  - new to voice telecommunications,
  - will be introduced by SP's when they deem necessary,
  - are required <u>end-to-end</u>
  - BUT improve international call quality considerably
- Universal use end-to-end seems a few years away yet…..

Low Bit Rate Codecs – Packet Transmission Overheads

- LBR codecs (narrow band codecs with Low Bit Rate)
  - low transmission rates are required when bandwidth is scarce or expensive.
  - packet overheads increase transmission rate (cost) but <u>do not</u> contribute to voice quality
  - reducing voice payload size to accommodate packet overheads reduces voice quality (NOT a recommended design approach)

Transmission "efficiency" (TDM = 100%)	Transmission rate	POS 6 bytes	IP 20 bytes	UDP 8 bytes	RTP 12 bytes	Voice Payload	
77%	82 kbit/s					160 bytes	G.711 64kbit/s / 20ms
30%	26.4 kbit/s					20 bytes	G.729 8kbits/s/ 20ms
46%	17.2 kbit/s					40 bytes	G.729 8kbit/s/ 40ms
34%	18.6 kbit/s					24 bytes	G.723.1 6.3kbit/s / 30ms
39%	17.6 kbit/s					30 bytes	GSM-EFR/AMR 12.2kbit/s / 20ms
57%	21.4 kbit/s					60 bytes	GSM-EFR/AMR 12.2kbit/s / 40ms

Low Bandwidth Codecs – Packet Transmission Overheads reduced by IP/UDP/RTP compression

- IP/UDP/RTP headers total 40 bytes
- IP/UDP/RTP compression
  - RFC 2508 reduces this to 2 bytes,
  - RFC 3095 Robust Header Compression (ROHC) reduces this to 1 byte
- Implementable only on a single hop

Transmission "efficiency" (TDM = 100%)	Transmission rate	POS 6 bytes	IP/UDP/RTP ROHC 1 byte	Voice Payload	
96%	66.8kbit/s			160 bytes	G.711 64kbit/s / 20ms
74%	10.8kbit/s			20 bytes	G.729 8kbit/s / 20ms
85%	9.4kbit/s			40 bytes	G.729 8kbit/s / 40ms
77%	8.1kbit/s			24 bytes	G.723.1 6.3kbit/s / 30ms
81%	15kbit/s			30 bytes	GSM-EFR/AMR 12.2kbit/s / 20ms
89%	13.6kbit/s			60 bytes	GSM-EFR/AMR 12.2kbit/s / 40ms

Low Bandwidth Codecs – Voice Transmission Overheads reduced by not transmitting silence

- Voice conversations have silent periods (while listening)
- Why transmit silences?
  - fill the gaps with “comfort noise”
    - comfort noise is generated by dynamic reconstruction of background noise from transmission of very low bandwidth noise characterisation signal
  - called Voice Activity Detection /Discontinuous Transmission (VAD/DTX)
  - saves Bandwidth with trivial quality impact
- Typical speech activity factors of 50% are obtained with normal speech, approximately halving the transmission bandwidth

Low Bandwidth Codecs – Transmission bandwidth

Bandwidth comparisons

Codec and DCME characteristics	Bandwidth per voice channel on SDH bearer (kbit/s)				
	TDM	VoIP			
		No IP/UDP/RTP compression		With IP/UDP/RTP compression	
		no VAD/DTX	with VAD/DTX*	no VAD/DTX*	with VAD/DTX*
G.711 64kbps	64	N/A	N/A	N/A	N/A
DCME, G.726 32kbps + VAD/DTX*	16	N/A	N/A	N/A	N/A
DCME, G.728 16kbps+ VAD/DTX*	8	N/A	N/A	N/A	N/A
Codec and Packetisation Period					
G.711 64kbps / 20ms	N/A	82	43	66.8	34.4
G.729 8kbps / 20ms	N/A	26.4	15	10.8	6.4
G.729 8kbps / 40ms	N/A	17.2	11	9.4	5.6
G.723.1 6.3kbit/s / 30ms	N/A	18.6	11.5	8.1	~5.0
AMR 12.2kbit/s / 20ms	N/A	31	17.6	15	8.4
AMR 12.2kbit/s / 40ms	N/A	21.4	13	13.6	7.6

**50% Speech Activity assumed*

Low Bandwidth Codecs – Transmission Cost

Cost Comparisons

Codec and DCME characteristics	TDM	VoIP: No IP/UDP/RTP compression, no VAD/DTX	VoIP: No IP/UDP/RTP compression, with VAD/DTX*	VoIP: With IP/UDP/RTP compression, no VAD/DTX	VoIP: With IP/UDP/RTP compression, with VAD/DTX*
Monthly cost of voice channel on SDH bearer (USD) based on INTELSAT Std B antenna, Global Beam, 2Mbit/s IDR with 3/4 FEC, 5yr tariff					
G.711 64kbps	$630	N/A		N/A	N/A
DCME, G.726 32kbps + VAD/DTX*	$157	N/A		N/A	N/A
DCME, G.728 16kbps+ VAD/DTX*	$79	N/A		N/A	N/A
Codec and Packetisation Period					
G.711 64kbps / 20ms	N/A	$807	$423	$657	$338
G.729 8kbps / 20ms	N/A	$260	$148	$106	$63
G.729 8kbps / 40ms	N/A	$169	$108	$92	$55
G.723.1 6.3kbit/s / 30ms	N/A	$183	$113	$80	$49
AMR 12.2kbit/s / 20ms	N/A	$305	$173	$148	$83
AMR 12.2kbit/s / 40ms	N/A	$210	$128	$134	$75

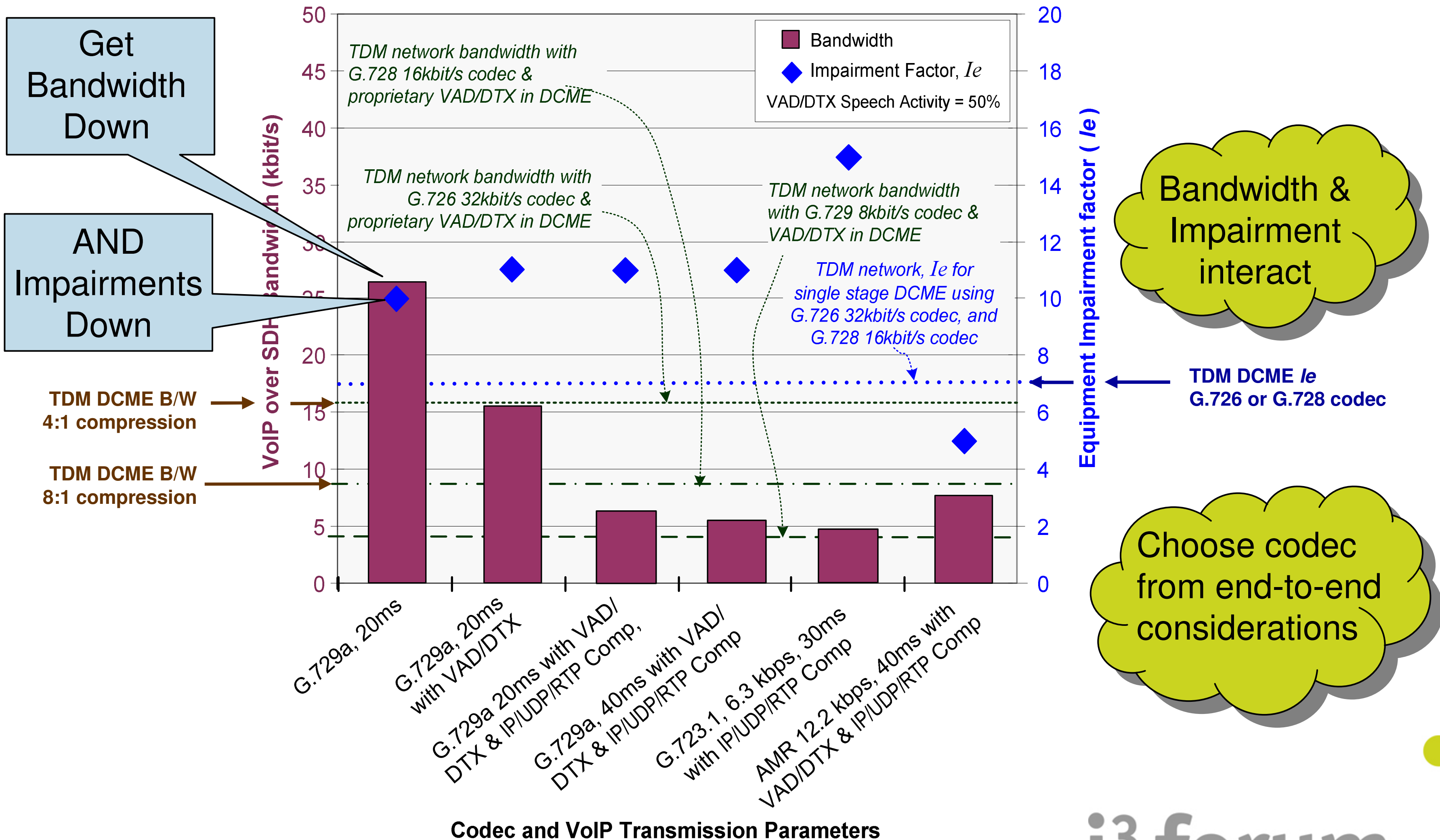
**50% Speech Activity assumed*

Finance Depts are used to these numbers

It is possible to engineer VoIP with similar costs

Low Bandwidth Codecs – Voice Call Quality/Bandwidth Tradeoff

Select LBR codec and transmission parameters with regard to end-end quality



Conclusions

- Narrow Band VoIP networks will provide lower quality international voice calls than TDM networks
- quality (all-cable) network calls to fall from "*Users Satisfied*" (regardless of international distance) to
  - intra region (e.g. Europe)
    - "*Users Satisfied*" without transcoding
    - "*Some/Many Users Dissatisfied*" with transcoding
  - long international calls such as New Zealand/Australia to UK/Europe
    - "*Some/Many Users Dissatisfied*" without transcoding
    - "*Nearly All Users Dissatisfied*" with transcoding
- end-to-end planning required -
  - direct bilateral interconnections will offer more predictable quality
  - multiple downstream networks will generally present quality difficulty
- Longer term, wideband codecs
  - potentially compensate for quality lost in transcoding
  - their introduction by SP's should be encouraged

Codec and Transmission Choices

- select codecs with low algorithmic latency.
- choose shorter packetisation periods
- keep packet loss as low as possible (< 0,1%)
- use Packet Loss Concealment whenever possible
- Avoid G.723.1 codec
- G.729 codec family offers a good balance of latency, bandwidth (cost) and voice quality
- Using the AMR codec in fixed networks would eliminate some transcoding impairments between fixed and mobile networks when serving Mobile SP's
- where occupied bandwidth is a critical cost parameter (satellite transmission)
  - select codecs with low bit rate and low *Ie* (balancing cost and voice quality end-to-end),
  - use Voice Activity Detection and Discontinuous Transmission (VAD/DTX),
  - consider translating packetisation period to higher values, such as 40ms,
  - implement IP/UDP/RTP compression

Transcoding

- Transcoding should be avoided unless absolutely necessary
- Many carriers end-to-end are likely to result in transcoding,
  - May render call quality completely unacceptable
  - Or even unintelligible
- Cooperation of all carriers and Service Providers in the call path will help maintain voice quality
- International Carriers should NOT transcode to save costs,
  - honour SP's codec choice where possible
  - different call Bandwidths could be tariffed differently
  - Satellite appears the exception where LBR codecs (PLUS other transmission techniques) appear necessary to justify costs
- Carrier/SP/Vendor cooperation is needed to achieve voice quality

Thank you!

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