Digital Signals and Testing
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Agenda and Objectives

- Testing downstream digital channels
- Test upstream performance
- VoIP Testing
QAM - Quadrature Amplitude Modulation

- Two carriers
  - Same frequency
  - 90° out-of-phase
  - Transmitted at the same time

- Multiple levels of amplitude & phase modulation

- Each carrier represents half the transmitted symbol.
QAM (Cont.)

- If I and Q channels transmit 4 levels of data per carrier
  - 16 symbols transmitted in one clock cycle
  - Each symbol contains 4 bits
  - Called 16QAM
- 8 levels per carrier
  - 64 symbols transmitted
  - Symbol contains 6 bits
  - 64QAM
- 16 levels per carrier
  - 256 symbols transmitted
  - Symbol contains 8 bits
  - 256QAM
Vectors & 16 QAM

Q 90°

Q 270°

I 180°

I 0°

00 01 10 11

0000 0001 0100 0101

1011 1010 1110 1111

01 01 01 01
Vectors and 16 QAM

1011

I 180°

Q 270°

Q 90°
### 64 QAM Constellation

**6 Bits per Symbol**

**QAM Analyzer Parameters**

- **ATT:** 20 dB
- **64 QAM**
- **BER (Pre-Fec):** 0
- **BER (Post-Fec):** 0
- **MER:** 34.1 dB
- **ENM:** 12.0 dB
- **FM:** 1.2%
- **ES:** 0 Sec
- **SES:** 0 Sec
- **FLS:** 0 Sec
- **UNAV:** 0 Sec
- **Elapsed:** 00:00:25
- **FILTER:** 4000
- **SYMB:** LOCK
- **FEC:** LOCK
- **STREAM:** LOCK

**Channel Power**

**Refresh Const.**

**Set QAM Parameters**

**More 1/2**

**General Information**

- **CF:** 573.000 MHz
- **Real Symbol:** 5.056 MS/s
- **SES THR:** 5.0E-03
- **=> Const. Stat. Equal. Analysis**
256 QAM

BER (Pre-Fec) 3.3E-07
BER (Post-Fec) 0
MER: 34.0 dB
ENM: 5.8 dB
EVM: 1.2 %
ES: 0 Sec
SES: 0 Sec
FLS: 0 Sec
UNAV: 0 Sec

QAM ANALYZER
CF: 567.000 MHz
Real Symbol: 5.360 MS/s
ATT: 20 dB

8 Bits per Symbol
Digital Measurements

- Digital Channel Power
- MER, ENM, and EVM
- Constellation Impairments
- Pre and Post FEC BER
- Adaptive Equalizer
- QIA Measurements
Analog vs. Digital Power Measurements

- 6 MHz
- 300 KHz
Digital Power Measurement

ATT: 10 dB  OFS: 0 dB

CF: 573.000 MHz
SPAN: 20 MHz

CHANNEL POWER: 6.7 dBmV = -42.0 dBm
AVERAGE 2
Balancing System Levels

DIGITAL CH POWER

ATT: 0 dB  OFS: 0 dB

BW: 6.00 MHz

CF: 741.000 MHz

SPAN: 20 MHz

FILE MANAGER

EDIT RECORDS ID

EDIT COMMENT

CHANNEL POWER: -16.0 dBmV = -64.7 dBm

AVERAGE 2
Modulation Error Ratio

- **MER** - Figure of Merit for quality of digital carriers

- CCN, CSO, CTB, laser compression, etc.... reduce MER (sum of all evils)

- 256 QAM picture tiles at 28dB MER

- Minimally good MER is 31 dB for 256 QAM at customer device
Modulation Error Ratio

- MER is defined as follows:
- MER is expressed in dB.

\[
\text{RMS error magnitude} = 10 \log \frac{\text{RMS Error Magnitude}}{\text{Average Symbol Magnitude}}
\]
Acceptable MER

- Output of QAM Modulator: 40 dB
- Input to Lasers: 39 dB
- Output of Nodes: 37 dB
- Output of Subscriber Taps: 35 dB
- At the input to the subscriber’s receiver: 34 dB
- Absolute minimum: 31 dB
MER and a Constellation

QAM Analyzer

- ATT: 20 dB
- 64 QAM
- CF: 573.000 MHz
- Real Symbol: 5.056 Ms/s
- SES THR: 5.0E-03
- => CONST. STAT. EQUAL. ANALYSIS

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
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<tbody>
<tr>
<td>BER (Pre-Fec)</td>
<td>0</td>
</tr>
<tr>
<td>BER (Post-Fec)</td>
<td>0</td>
</tr>
<tr>
<td>MER</td>
<td>34.1 dB</td>
</tr>
<tr>
<td>ENM</td>
<td>12.0 dB</td>
</tr>
<tr>
<td>EVM</td>
<td>1.2 %</td>
</tr>
<tr>
<td>ES</td>
<td>0 Sec</td>
</tr>
<tr>
<td>SES</td>
<td>0 Sec</td>
</tr>
<tr>
<td>FLS</td>
<td>0 Sec</td>
</tr>
<tr>
<td>UNAV</td>
<td>0 Sec</td>
</tr>
</tbody>
</table>

Elapsed: 00:00:25  FILTER: 4000
SYMB: LOCK  FEC: LOCK  STREAM: LOCK

Digital Channel Power

Refresh Const.

Set QAM Parameters

More 1/2
Constellation Analysis
64 QAM With Gain Compression
Constellation Showing Signal Noise
Coherent Distortions, or Interference
<p>| | | | | |</p>
<table>
<thead>
<tr>
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</tr>
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</table>

64 QAM With I/Q Gain Imbalance
Constellation Showing Phase Noise
What is BER?

**Definition:**
BER is defined as the ratio of the number of wrong bits over the number of total bits.

\[
\text{Sent Bits} \quad 1101101101 \\
\text{Received Bits} \quad 1100101101
\]

\[
\text{BER} = \frac{\# \text{ of Wrong Bits}}{\# \text{ of Total Bits}} = \frac{1}{10} = 0.1
\]
What is BER?

- BER displayed in Scientific Notation.
- The more negative the exponent, the better the BER.
- Pre-FEC must be Better than 1.0E-6

<table>
<thead>
<tr>
<th>Fraction</th>
<th>Decimal</th>
<th>Scientific Notation</th>
</tr>
</thead>
<tbody>
<tr>
<td>1/1</td>
<td>1</td>
<td>1.0E+00</td>
</tr>
<tr>
<td>1/10</td>
<td>0.1</td>
<td>1.0E-01</td>
</tr>
<tr>
<td>1/100</td>
<td>0.01</td>
<td>1.0E-02</td>
</tr>
<tr>
<td>1/1,000</td>
<td>0.001</td>
<td>1.0E-03</td>
</tr>
<tr>
<td>1/10,000</td>
<td>0.0001</td>
<td>1.0E-04</td>
</tr>
<tr>
<td>1/100,000</td>
<td>0.00001</td>
<td>1.0E-05</td>
</tr>
<tr>
<td>1/1,000,000</td>
<td>0.000001</td>
<td>1.0E-06</td>
</tr>
<tr>
<td>1/10,000,000</td>
<td>0.0000001</td>
<td>1.0E-07</td>
</tr>
<tr>
<td>1/100,000,000</td>
<td>0.00000001</td>
<td>1.0E-08</td>
</tr>
<tr>
<td>1/1,000,000,000</td>
<td>0.000000001</td>
<td>1.0E-09</td>
</tr>
<tr>
<td>2/1,000</td>
<td>0.002</td>
<td>2.0E-03</td>
</tr>
</tbody>
</table>
Pre and Post FEC errors

- **Pre FEC errors**
  - Errors measured before the FEC (Forward Error Correction)

- **Post FEC errors**
  - Errors that could not be corrected

- Devices tolerate pre-FEC errors up to \textit{1E-06 or one error in one million bits}. After that the FEC can do no more.

- **Post-FEC errors**
  - Retransmissions requests, and slowdowns in a DOCSIS systems
  - Tiling in Digital Video
Pre and Post FEC BER

- Need to know pre and post FEC bit error rate.
- FEC decoder needs Pre BER better than $1 \times 10^{-6}$
- Post FEC Bit errors are \textit{not acceptable}.
BER and a Constellation

- BER (Pre-Fec): 0
- BER (Post-Fec): 0

- ATT: 20 dB  64 QAM
- CF: 573.000 MHz  Real Symbol: 5.056 MS/s

- MER: 34.1 dB
- ENM: 12.0 dB
- EVM: 1.2 %
- ES: 0 Sec
- SES: 0 Sec
- FLS: 0 Sec
- UNAV: 0 Sec

Elapsed: 00:00:25  FILTER: 4000
SYMB: LOCK  FEC: LOCK  STREAM: LOCK

=> CONST. STAT. EQUAL. ANALYSIS
DIGITAL CHANNEL POWER
REFRESH CONST.
SET QAM PARAMETERS
MORE 1/2
Statistical Mode

- ATT: 60 dB
- QAM: 16
- PL: 0.00000%
- CF: 63.000 MHz
- Real Symbol: 2.560 MS/s
- CONT: 5.0E-03
- EQUAL.
- ANALYSIS
- DIGITAL
- CHANNEL
- POWER
- STAT.
- PERIOD
- START
- => STOP
- => STAT.
- SET
- QAM
- PARAMETERS
- MORE
- 1/2
- Elapsed 01:00 of 1 (Min)
- SYMB: LOCK
- FEC: LOCK
- STREAM: LOCK

- BER (Pre-Fec)
  - AVG: 4.0E-08
- BER (Post-Fec)
  - AVG: 0
- MER: 38.1 dB
- ENM: 9.9 dB
- EVM: 0.7 %
- ES: 0 Sec
- SES: 0 Sec
- FLS: 0 Sec
- UNAV: 0 Sec
Adaptive Equalizer

- Every MPEG2 digital receiver uses an Adaptive Equalizer

- Performs 3 functions
  - Compensates for amplitude imperfections
  - Compensates for group delay
  - Rings at the symbol rate to only allow one symbol at a time into the digital receiver
Equalizer Mode

ATT: 35 dB
J83ANNEX B
CF: 603.000 MHz
SES Threshold: 5.0e-003
Real Symbol: 5.361 Ms/s

BER (Pre-Fec)
0.0e-12
BER (Post-Fec)
0.0e-12
MER: > 40.0 dB
ENM: 11.8 dB
EVM: < 0.8 %
ES: 0 Sec
SES: 0 Sec
FLS: 0 Sec
UNAV: 0 Sec

Elapsed: 00:00:00
Symb: Lock  Fec: Lock  Stream: Lock

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Amplitude Ripple

An in-service spectrum analyzer measurement
Group Delay

- Definition:
  - Group delay is a measure of how long it takes a signal to traverse a network, or its transit time. It is a strong function of the length of the network, and usually a weak function of frequency. It is expressed in units of time, pico-seconds for short distances or nanoseconds for longer distances.

- Measured in units of time,
  - Typically nanoseconds (ns) over frequency
  - Or, Delay per MHz.

- Ideal system
  - All frequencies move through the system with equal time delay

- Frequency response problems increase group delay

- Group delay is worse near band edges and diplex filter roll-off areas
Group Delay (cont’d)

- Excessive group delay increases bit error rate due to intersymbol interference

- DOCSIS spec.
  - no greater than 200nSecs per MHZ
  - Spec should be less than 100 nSecs per MHz
As different frequencies pass through a Cable System, some will move faster than others.
Group Delay Measurement

ATT: 35 dB  Modulation: QAM256  SES Threshold: 5.0e-003
J83ANNEX B  CF: 603.000 MHz  Real Symbol: 5.361 MS/s

<table>
<thead>
<tr>
<th>BER (Pre-Fec)</th>
<th>0.0e-12</th>
</tr>
</thead>
<tbody>
<tr>
<td>BER (Post-Fec)</td>
<td>0.0e-12</td>
</tr>
<tr>
<td>MER:</td>
<td>&gt; 40.0 dB</td>
</tr>
<tr>
<td>ENM:</td>
<td>11.8 dB</td>
</tr>
<tr>
<td>EVM:</td>
<td>&lt; 0.6 %</td>
</tr>
<tr>
<td>ES:</td>
<td>0 Sec</td>
</tr>
<tr>
<td>SES:</td>
<td>0 Sec</td>
</tr>
<tr>
<td>FLS:</td>
<td>0 Sec</td>
</tr>
<tr>
<td>UNAV:</td>
<td>0 Sec</td>
</tr>
</tbody>
</table>

Elapsed: 00:00:00
SYMB: LOCK  FEC: LOCK  STREAM: LOCK

V1: 603.300 MHz  -21.0 ns
V2: 603.984 MHz  13.0 ns
dV: 0.684 MHz   34.0 ns
49.7 ns/MHz
QAM Impairment Analysis (QIA)

- **I/Q Gain and Phase**
  - Phase & gain of the I & Q carriers must be equal in order for the constellation to be correct.
  - Impairment is caused by the QAM modulators.
  - Gain difference between the 2 carriers should be less than 1.8%
  - Phase difference should be less than 1 degree.

- **Echo Margin**
  - Measurement in dB of the distance of the equalizer taps are from the template
  - Caused by impedance mismatches in the system.
  - Should be at least 6 dB.
QIA Continued

- **Carrier Offset**
  - Carrier frequency test.
  - Should be no more than +/- 25KHz

- **Estimated Noise Margin**
  - Difference in dB between MER & the digital cliff
  - Digital cliff dependent on modulation type; 64 or 256 QAM
  - Example if the Minimum MER for 256 QAM is 28 and the measurement is 34, than the ENM is 6

- **Frequency Response**
  - Should be less than 3 dB pk-to-pk

- **Hum**
  - Low frequency disturbances
  - Same as hum on analog carriers, if the level is the same, it’s the system, if higher on the digitals then it’s probably the QAM modulator
QIA Continued

- Symbol Rate Error
  - Should be less than +/- 5 ppm

- Phase Noise
  - Variation is phase of the carriers
  - Typically caused by signal going through a frequency conversion as in an up-converter
  - Should be less than .5 degrees

- Group Delay
  - Should be less than 50 nSec pk-to-pk

- Compression
  - Caused by overdriving lasers or amplifiers
  - Shows up as corners pulling in at the outer corners of the constellation
  - Should less than 1%
Going to 16 QAM in the upstream

- QPSK carriers are strictly phase modulated.
- Most ingress does not affect phase.
- Amplitude modulation is much more susceptible to ingress
- 16 QAM carriers are amplitude and phase modulated
- 16 QAM is less robust because it is a higher order of QAM modulation
Upstream 16 QAM Testing

Before moving to 16 QAM you should check:

- Spectrum analysis of the upstream

- Compression of the return laser due to added carrier or a carrier with added bandwidth

- Group Delay of the new carrier

- MER and BER of the new carrier

- Amplitude Ripple

- Micro-reflections
Testing 16 QAM in the upstream

- **Recommended Method:**
  - Inject a 16 QAM carrier from the field
  - Analyze performance at the headend or hub.

- **Recommended Measurements:**
  - Spectrum analysis for Ingress
  - Constellation
  - MER
  - BER
  - Frequency response
  - Group Delay
Upstream Spectrum
Upstream Spectrum Display Showing Clipping
A Good 16 QAM Constellation

- ATT: 30 dB
- Modulation: QAM16
- CF: 38.000 MHz
- SES Threshold: 1.0e-002
- Real Symbol: 2.560 MS/s

- BER (Post-Fec): 0.0e-12
- MER: 38.9 dB
- ENM: 10.7 dB
- EVM: 0.8 %
- ES: 0 Sec
- SES: 0 Sec
- FLS: 0 Sec
- UNAV: 0 Sec

Elapsed: 00:00:00
Sample: 8192
Sym: LOCK
FEC: LOCK
Stream: LOCK
16 QAM Constellation with Compression

- ATT: 15 dB
- Modulation: QAM16
- SES Threshold: 5.0e-003
- CF: 40.800 MHz
- Real Symbol: 2.560 MS/s

- BER (Pre-Fec): 0.0e-12
- BER (Post-Fec): 0.0e-12
- MER: 26.5 dB
- ENM: < 1.0 dB
- EVM: 2.9%
- ES: 0 Sec
- SES: 0 Sec
- FLS: 0 Sec
- UNAV: 0 Sec

Elapsed: 00:00:00  SAMPLE: 8192
EXMD: LOCK  EES: LOCK  STREAM: LOCK

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Microreflections

- Are impedance mismatches
- Impedance is nominal at best
- Impedance mismatches are everywhere:
  - connectors, amplifiers inputs and outputs, passive device inputs and outputs, and even the cable itself
- More affect on Upstream
  - Attenuation is lower than downstream, therefore upstream micro-reflections tend to be worse
- Every mismatch reflects some energy back toward the source
Microreflections

- Affect higher orders of modulation more
  - 16 QAM is affected more than QPSK

- Adaptive equalizer compensates for downstream micro-reflections

- Adaptive equalizer may not be used on Upstream

- NOTE: Pre-equalization can be used on Upstream
Microreflections

Causes:

- Damaged or missing end-of-line terminators
- Damaged or missing chassis terminators on directional coupler, splitter, or multiple-output amplifier unused ports
- Loose center conductor seizure screws
- Unused tap ports not terminated—this is especially critical on low value taps
Microreflections

Causes (cont’d):

● Use of so-called self-terminating taps at feeder ends-of-line

● Kinked or damaged cable (includes cracked cable, which causes a reflection and ingress)

● Defective or damaged actives or passives (water-damaged, water-filled, cold solder joint, corrosion, loose circuit board screws, etc.)

● Cable-ready TVs and VCRs connected directly to the drop (return loss on most cable-ready devices is poor)

● Some traps and filters have been found to have poor return loss in the upstream, especially those used for data-only service
- Here’s an example: An approx. -33 dBc echo at just over 1 µsec
- This echo meets the DOCSIS upstream -30 dBc at >1.0 µsec parameter
- Here, the echo is sufficient to cause some amplitude and group delay ripple
This example shows about 40 nanoseconds of in-channel group delay ripple.
Group Delay Measurement

- ATT: 3 dB
- Modulation: QAM256
- SES Threshold: 5.0e-003
- Real Symbol: 5.120 MS/s

- BER (Pre-Fec): 0.0e-12
- BER (Post-Fec): 0.0e-12
- MER: 31.6 dB
- ENM: 3.4 dB
- EVM: 1.6%
- ES: 0 Sec
- SES: 0 Sec
- FLS: 94 Sec
- UNAV: 0 Sec

Group Delay

- Elapsed: 00:00:00
- EXMR: LOCK
- E56: UPLINK
- STREAM: UPLINK

- V1: 11.228 MHz, 67.0 ns
- V2: 10.883 MHz, 68.0 ns
- dV: 0.345 MHz, 135.0 ns, 390.9 ns/MHz
Amplitude Ripple

An in-service spectrum analyzer measurement
Frequency Response of an Upstream Carrier

ATT: 5 dB  Modulation: QAM256  SES Threshold: 5.0e-003
J03ANNEX B  CF: 10.500 MHz  Real Symbol: 5.120 MS/s

BER (Pre-Fec)
0.0e-12

BER (Post-Fec)
0.0e-12

MER: 31.5 dB
ENM: 3.3 dB
EVM: 1.6 %

ES: 0 Sec
SES: 0 Sec

FLS: 31 Sec
UNAV: 0 Sec

Frequency Response

Elapsed: 00:00:00

SMP: LOCK  FSC: UNLOCK  STREAM: UNLOCK
Conclusions

- Digital signal quality
  - MER
  - Pre and Post BER

- A statistical mode can be used to measure over time.

- Adaptive equalizers that are present in digital receivers can also tell us much about the incoming signals.

- The upstream can be qualified by injecting a 16 QAM signal and making the same measurements used in the downstream.
VOIP Measurements
Two Categories of VOIP Testing

1. Network Components
   - Latency
   - Jitter
   - Lost Packets

2. Voice Quality
   - Mean Opinion Score
   - R Factor
Voice traffic must be digitized, optionally compressed, processed for echo canceling, and packetized.

Voice packets may take multiple hops.

Result is that voice over IP networks has more delay than traditional circuit switched approaches.

Solutions

- Run with very small packetization period (10ms).
- Minimize processing delays in system components.
- Minimize number of hops from source to destination.
- Give voice packets priority.
Latency & Jitter

- Latency and Jitter cause voice quality degradation
- Both can be measured using the UGS service flow carrying ICMP ping packets
Latency Measurement

- Latency is simply the transit time between one network element to another
- End to end delay

LATENCY = TIME
Directing ping packets through a UGS pipe (service flow) provides a good platform for testing latency to the network side of the CMTS.
VoIP Issue - Jitter

- Delay in routers varies with current traffic load.
- Voice packets can take different routes.
- Packets don’t always arrive in the order they were sent.
- Net result is variability in delay known as jitter.

Solution.

- Jitter buffer at the playout side.
- Received packets are placed here first before playout.
- Builds in a standard delay to allow packets to arrive, get buffered up, then played out.
- User hears no gaps between delayed packets.
- Give voice packets priority.
Jitter = Change in Latency

- Transmitted packets evenly spaced in time
- Received packets unevenly spaced in time
- Packets are experiencing changes in transit time (Latency)
- Delta Latency = Jitter
- Buffering is used to compensate for jitter, but buffering can contribute to latency
VoIP Issue - Lost Packets

- There is no retransmission!
- Solutions – PLC Packet Loss Concealment
  - If single packet is lost, replay the last packet because the human ear is very forgiving.
  - After one replay, play white noise.
VOIP & Lost Packets

- **VOIP** is very *sensitive to lost packets*

- A 1% packet loss causes **call break up**

- A 3% packet loss will **drop the call** completely

- Telcos spec 0.5% packet loss or better for High Speed data/voice circuits
What’s Good? What’s bad?

- **Latency**
  - Preferred <60 mSec
  - Maximum 150 mSec
  - Very annoying about 250 mSec

- **Jitter**
  - Preferred equal to or <20 mSec
  - Maximum should be <30 mSec
## VOIP Network Test Results

### DS/US Measurements
- **Lost Packets**
- **Jitter**
- **Latency**

### VoIP Details

<table>
<thead>
<tr>
<th>Category</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>DOCSIS Mode</strong></td>
<td>1.1</td>
</tr>
<tr>
<td><strong>Security Mode</strong></td>
<td>BPI+</td>
</tr>
<tr>
<td><strong>QoS Class</strong></td>
<td>3 Platinum</td>
</tr>
<tr>
<td><strong>Rx Level</strong></td>
<td>10.5 dBmV</td>
</tr>
<tr>
<td><strong>MER</strong></td>
<td>36 dB</td>
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<tr>
<td><strong>PreBER</strong></td>
<td>1.0E-8</td>
</tr>
<tr>
<td><strong>PstBER</strong></td>
<td>1.0E-9</td>
</tr>
<tr>
<td><strong>Freq.</strong></td>
<td>747.000 MHz</td>
</tr>
<tr>
<td><strong>Mod.</strong></td>
<td>256 QAM</td>
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<tr>
<td><strong>Tx Level</strong></td>
<td>35.5 dBmV</td>
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<tr>
<td><strong>Lost Pkts</strong></td>
<td>00012</td>
</tr>
<tr>
<td><strong>Disc. Pkts</strong></td>
<td>00006</td>
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<tr>
<td><strong>% Lost Pkts</strong></td>
<td>0.5 %</td>
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<tr>
<td><strong>Latency</strong></td>
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<tr>
<td><strong>Jitter</strong></td>
<td>&lt;5 mSec</td>
</tr>
</tbody>
</table>
Voice Quality Tests

- **MOS** – Mean Opinion Score
  - Actual listeners grade performance on a scale of 1-5

- We now use a MOS server and send voice data packets to the server for analysis.

- The server also sends voice data back to the instrument which also performs analysis.

- This both upstream and downstream are analyzed separately.
Factors Affecting Voice Quality

- Latency, Jitter and Lost Packets
- Noise
- Delay
- Echo
- Intelligibility
What’s Acceptable

- MOS minimum 3.1, good 3.5
- R Factor minimum 70%, good 80%
Thank You!
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